Ref #	Hits	Search Query	DBs	Default Operator	Plurals	Time Stamp
L1	170849	(data bit) near2 rate	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 08:06
L2	7762	L1 same ((mobile base) near (station unit))	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 08:07
L3	4236	l2 and cdma	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 08:06
L4	340	repeat$3 same punctur$3 same transmi$6	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 08:08
L5	104	l2 and L4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 09:35
L6	2403	709/245	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 09:36
L7	1998	709/246	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 09:36
L8	1268	cdma and punctur$3	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 09:36
L9	1	l6 and L8	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 09:38
L10	0	l7 and L8	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 09:38

L11	1	(l6 l7) and l4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 09:39
L12	478	(l6 l7) and (qos (quality near3 service))	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 09:40
L13	32	l12 and cdma	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 09:40
S1	1264	cdma and punctur$3	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 12:32
S2	333	S1 and (qos (quality near3 service))	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 14:52
S3	1	("6591150").pn.	US-PGPUB; USPAT; EPO; JPO; IBM_TDB	OR	ON	2005/04/01 07:56
S4	1264	cdma and punctur$3	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 10:50
S5	333	S4 and (qos (quality near3 service))	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 14:54
S6	101	S5 and (block near3 (size length))	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 09:37
S7	105	S5 and ("192" "384" "768" "1536" "2048" "3072")	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 10:05

S8	62	symbol near2 prun$3	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 10:15
S9	419	(455/452.2).CCLS.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2005/04/01 10:15
S10	1550	(455/450).CCLS.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2005/04/01 10:17
S11	2765	(370/342).CCLS.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2005/04/01 10:17
S12	636	(370/441).CCLS.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2005/04/01 10:17
S13	149	(370/542).CCLS.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2005/04/01 10:17
S14	55	(370/543).CCLS.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2005/04/01 10:18
S15	109	S12 and (qos (quality near3 service))	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 10:50
S16	4	("6088342" | "6317430" | "6507582" | "6553003").PN.	US-PGPUB; USPAT; USOCR	OR	ON	2005/04/01 10:26
S17	7	("5504773" | "5790534" | "5831978" | "6167270" | "6275712" | "6442152" | "6456604").PN.	US-PGPUB; USPAT; USOCR	OR	ON	2005/04/01 10:49
S18	109	S15 and S12	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 10:50

S19	22	S15 and S4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 10:50
S20	19	intra?media	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 12:37
S21	31	intra?media (intra adj media)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 12:38
S22	102	inter?media (inter adj media)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 12:37
S23	1293	punctur$3 with (repeat$3 repitition duplicat$5 redundan$3)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 14:53
S24	1062	punctur$3 with (repeat$3 repitition duplicat$5)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 15:06
S25	13	S24 same (qos (quality near3 service))	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 14:54
S26	233	S24 and cdma	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/01 15:06
S27	1	"22693"	EPO	OR	ON	2005/04/06 12:19
S28	4	("6,747,963" "6,606,341" "6,621, 809" "6,742,742").pn.	USPAT	OR	ON	2005/04/08 16:38
S29	6	("5566189" | "5828677" | "6014411" | "6385752" | "6421803" | "6519731").PN.	US-PGPUB; USPAT; USOCR	OR	ON	2005/04/06 13:44
S30	25	punctur$3 with (repeat$4 repitition) with (equal equivalent)	US-PGPUB; USPAT; USOCR	OR	ON	2005/04/06 13:55

S31	38	punctur$3 with (repeat$4 repitition) with (identical equal equivalent)	US-PGPUB; USPAT; USOCR	OR	ON	2005/04/06 13:55
S32	341	repeat$3 same punctur$3 same transmi$6	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 09:34
S33	18	("5309474" | "5490136" | "5581575" | "5822318" | "5909434" | "6181683" | "6347091" | "6374112" | "6438370" | "6456607" | "6501748" | "6501953" | "6507567" | "6577669" | "6594238" | "6680932" | "6697986" | "6697988").PN.	US-PGPUB; USPAT; USOCR	OR	ON	2005/04/07 09:43
S34	0	("6788657").URPN.	USPAT	OR	ON	2005/04/07 09:45
S35	6	("5566189" | "5828677" | "6014411" | "6385752" | "6421803" | "6519731").PN.	US-PGPUB; USPAT; USOCR	OR	ON	2005/04/07 09:51
S36	31571	(qos ((quality grade) near2 service))	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/07 16:03
S37	53	(qos ((quality grade) near2 service)) near4 (separat$5 segregat$5) near5 (stream packet data)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 15:28
S38	1413	(714/6).CCLS.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2005/04/07 17:03
S39	3	S38 and punctur$3	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/07 17:08
S40	1034	bit near2 rob$3	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/07 17:10
S41	1	S38 and S40	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/07 17:09

S42	225	bit near2 (rob robbing)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/07 17:10
S43	340	repeat$3 same punctur$3 same transmi$6	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 09:34
S44	167	S43 and turbo	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 10:26
S45	1467	RLP (radio adj link adj protocol)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 10:27
S46	115	S45 same (qos (quality near2 service) priority)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 15:26
S47	170849	(data bit) near2 rate	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/11 08:05
S48	1	S43 same S45 same S47	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 13:51
S49	1430	(data near2 rate near2 request) DRQ	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 13:52
S50	10	S49 and S43	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 13:54
S51	206	S45 same S47	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 13:55

S52	2	("5793744" | "5920576").PN.	US-PGPUB; USPAT; USOCR	OR	ON	2005/04/08 14:12
S53	3	S46 and (intra?media (intra adj1 media))	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 15:26
S54	285	multimedia near4 (separat$5 segregat$5) near5 (stream packet data)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 16:03
S55	3	multimedia near4 (separat$5 segregat$5) near5 (stream packet data) same (qos gos ((grade quality) near2 service))	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 16:18
S56	2336	(separat$5 segregat$5) near5 (stream packet data) same (qos gos ((grade quality) near2 service))	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 16:03
S57	6	S56 and S43	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 15:30
S58	1752	(separat$5 segregat$5) near3 (stream packet data) same (qos gos ((grade quality) near2 service))	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 16:03
S59	96	S58 and cdma	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 16:04
S60	40	application near4 (separat$5 segregat$5) near5 (stream packet data) same (qos gos ((grade quality) near2 service))	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/04/08 16:28
S61	6	("5674003"|"6119171"|"6141686"|"6141686"|"6286052"|"6377996"|"6714992").PN.	USPAT	OR	ON	2005/04/08 16:39

Terms used
puncture repeat grade quality service qos gos cdma

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21 A CDMA-based radio interface for third generation mobile systems
Sergio Barberis, Ermanno Berruto
June 1997 **Mobile Networks and Applications**, Volume 2 Issue 1
Full text available: pdf(257.24 KB)   Additional Information: full citation, abstract, references, index terms

This paper deals with the use of a CDMA-based radio interface in third generation mobile systems (Universal Mobile Telecommunications System—UMTS, and Future Public Land Mobile Telecommunications System—FPLMTS). The paper is not intended as a detailed analysis of the radio interface performance, but as an overview of the main issues arising in a typical CDMA-based mobile system, discussing the different available technical solutions. First of all, the basic requirements of the r ...

22 Impact of statistical multiplexing on voice quality in cellular networks
T. Enderes, S. C. Khoo, C. A. Somerville, K. Samaras
August 2000 **Proceedings of the 3rd ACM international workshop on Modeling, analysis and simulation of wireless and mobile systems**
Full text available: pdf(817.16 KB)   Additional Information: full citation, abstract, references, citings, index terms

This paper examines the quality of transmission of voice over cellular, packet-switched TDMA networks. The medium access mechanism in the uplink is simulated for different statistical multiplexing scenarios in order to assess the effect of front-end clipping on voice quality. Moreover, the simulation is implemented in a real-time demonstration platform utilized to acquire subjective indicators of voice quality by performing Mean Opinion Score (MOS) tests. The results from the MOS tests are ...

23 On the effects of adaptive forward error correction mechanism in direct broadcast satellite networks
Fatih Alagöz, David Walters, Amina Alrustamani, Branimir Vojcic, Raymond Pickholtz
August 1999 **Proceedings of the 2nd ACM international workshop on Modeling, analysis and simulation of wireless and mobile systems**
Full text available: pdf(877.12 KB)   Additional Information: full citation, references, citings, index terms

24 Impact of statistical multiplexing on voice quality in cellular networks
Thomas Enderes, Swee Chern Khoo, Clare A. Somerville, Kostas Samaras

April 2002 **Mobile Networks and Applications**, Volume 7 Issue 2

Full text available: pdf(239.13 KB) Additional Information: full citation, abstract, references, index terms

This paper examines the quality of transmission of voice over cellular, packet-switched networks. The medium access mechanism in the uplink is simulated under various statistical multiplexing scenarios in order to assess the effect of front-end clipping on voice quality. Moreover, the simulation is implemented in a real-time demonstration platform utilized to acquire subjective indicators of voice quality by performing Mean Opinion Score (MOS) tests. Results from the MOS tests are reported, and ...

Keywords: cellular networks, speech pattern, statistical multiplexing, voice quality

25 A comprehensive approach to signaling, transmission, and traffic management for wireless ATM networks

Anthony Burrell, P. Papantoni-Kazakos

September 2001 **Wireless Networks**, Volume 7 Issue 4

Full text available: pdf(331.30 KB) Additional Information: full citation, abstract, references, index terms

We propose and evaluate a signaling and transmission algorithmic system for wireless digital networks, in conjunction with a Traffic Monitoring Algorithm (TMA) for dynamic capacity allocation in multimedia ATM environments. The deployed signaling protocol is stable, and two transmission techniques are compared: a Framed Time-Domain Based (FTDB) technique and a Framed CDMA (FCDMA) technique. The overall signaling/transmission/traffic monitoring proposed system has powerful performance characteris ...

Keywords: multimedia environments, signaling and transmission, traffic monitoring, wireless

26 Performance aspects of data broadcast in wireless networks with user retrials

Natalija Vlajic, Charalambos D. Charalambous, Dimitrios Makrakis

August 2004 **IEEE/ACM Transactions on Networking (TON)**, Volume 12 Issue 4

Full text available: pdf(1.28 MB) Additional Information: full citation, abstract, references, index terms

The User retrial phenomenon and its significant impact on network performance in unicast wireless systems are known and relatively well studied in the literature. However, there have been no previous studies on the impact of the user retrial phenomenon on other types of wireless networks. The objective of this paper is to extend the analysis of the user retrial phenomenon to wireless systems which, in addition to unicast service, also support a data broadcast service.This objective is realized b ...

Keywords: data broadcast, mobile communication, wireless networks

27 Medium access control for ATM-to-CDMA interface

Aykut Hocanin, Shanuj V. Sarin, Hakan Deliç

July 2002 **Wireless Networks**, Volume 8 Issue 4

Full text available: pdf(180.82 KB) Additional Information: full citation, abstract, references, index terms

An integrated medium access control framework for a direct sequence code-division multiple access (CDMA) radio access to an asynchronous transfer mode (ATM) network is considered. The system accommodates multimedia services such as voice and data. The inherently high error rate associated with the multipath fading channel is partly overcome by the introduction of a data link control layer employing one- and two-dimensional CRC codes for error detection/correction in voice and data packets, respe ...

Keywords: ARQ, ATM, CDMA, CRC, multipath fading

28 A cellular wireless local area network with QoS guarantees for heterogeneous traffic
Sunghyun Choi, Kang G. Shin
June 1998 **Mobile Networks and Applications**, Volume 3 Issue 1

Full text available: pdf(374.40 KB) Additional Information: full citation, abstract, references, citings, index terms

A wireless local area network (WLAN) or a cell with quality-of-service (QoS) guarantees for various types of traffic is considered. A centralized (i.e., star) network is adopted as the topology of a cell which consists of a base station and a number of mobile clients. Dynamic Time Division Duplexed (TDD) transmission is used, and hence, the same frequency channel is time-shared for downlink and uplink transmissions under the dynamic control of the base station. We divide traffic into two cl ...

29 QoS provisioning in wireless/mobile multimedia networks using an adaptive framework
Taekyoung Kwon, Yanghee Choi, Chatschik Bisdikian, Mahmoud Naghshineh
January 2003 **Wireless Networks**, Volume 9 Issue 1

Full text available: pdf(188.34 KB) Additional Information: full citation, abstract, references, index terms

Recently there is a growing interest in the adaptive multimedia networking where the bandwidth of an ongoing multimedia call can be dynamically adjusted. In the wireless/mobile multimedia networks using the adaptive framework, the existing QoS provisioning focused on the call blocking probability and the forced termination probability should be modified. We, therefore, redefine a QoS parameter -- the *cell overload probability* -- from the viewpoint of the adaptive multimedia networking. Th ...

Keywords: QoS provisioning, adaptive framework, adaptive multimedia, bandwidth adaptation, call admission control, wireless/mobile multimedia network

30 Traffic and interference adaptive scheduling for internet traffic in UMTS
Marco Conti, Enrico Gregori
August 2004 **Mobile Networks and Applications**, Volume 9 Issue 4

Full text available: pdf(316.53 KB) Additional Information: full citation, abstract, references, index terms

In this paper we propose a scheduling strategy for the radio resources management when transmitting Internet traffic over third-generation systems. More precisely, we consider the UMTS Terrestrial Radio Access Network (UTRAN) Time Division Duplex (TDD) mode standardized by ETSI. UTRAN TDD uses a hybrid solution of code and time division multiple access, called TD-CDMA. In UMTS systems a key issue in developing access methodologies for the available spectrum is an optimal management of the rare r ...

Keywords: UMTS, UTRA-TDD, internet traffic, scheduling

31 On admission control and scheduling of multimedia burst data for CDMA systems
Yu-Kwong Kwok, Vincent K. N. Lau
September 2002 **Wireless Networks**, Volume 8 Issue 5

Full text available: pdf(223.56 KB) Additional Information: full citation, abstract, references, citings, index terms

In order to support transmissions of multimedia data (high data rate and burst) with performance guarantees in a wideband CDMA system, it is crucial to design a judicious

algorithm for burst data admission control and scheduling. However, in the current literature there are only simple techniques (such as first-come-first-served and equal sharing) suggested for tackling the problem. Indeed, these existing schemes are not designed for optimizing the precious bandwidth resources while providing pe ...

Keywords: 3G, CDMA, admission control, burst data, cdma2000, integer programming, optimal algorithm

32 Issues in satellite personal communication systems
Erich Lutz
February 1998 **Wireless Networks**, Volume 4 Issue 2

Full text available: pdf(742.57 KB) Additional Information: full citation, abstract, references, citings, index terms

In the paper various issues in personal satellite communications are addressed. Basic geostationary and non-geostationary satellite constellations are considered. The narrowband and wideband characterization of the mobile satellite channel and related system implications are discussed. Satellite diversity is presented as a measure to overcome signal shadowing. The capacity of TDMA and CDMA multiple access is estimated, taking into account co-channel interference. Various network issues, suc ...

33 Reverse-link capacity of multiband overlaid DS-CDMA systems
Lei Zhuge, Victor O. K. Li
April 2002 **Mobile Networks and Applications**, Volume 7 Issue 2

Full text available: pdf(245.27 KB) Additional Information: full citation, abstract, references, citings, index terms

In the future deployment of wideband DS-CDMA mobile communication systems, spectrum overlay among sub-bands with different bandwidths is probably inevitable. In this paper we present an approach to estimate the reverse-link capacity of overlaid multiband DS-CDMA systems in terms of the maximum number of users in each sub-band. We will derive the general capacity formula, and present a decomposition method for the capacity analysis to reduce the computational complexity. Based on this decompositi ...

Keywords: multiband overlaid DS-CDMA, reverse-link capacity, wireless multimedia systems

34 Admission control with priorities: approaces for multi-rate wireless system
Deepak Ayyagari, Anthony Ephremides
October 1999 **Mobile Networks and Applications**, Volume 4 Issue 3

Full text available: pdf(147.14 KB) Additional Information: full citation, abstract, references, citings, index terms

Priority based link-bandwidth partitioning is required to support wireless multimedia services, having diverse QoS (delay, throughput) requirements, in mobile ad hoc networks with multimedia nodes. A new class of service disciplines, termed "batch and prioritize" or BP admission control (AC), is proposed. The BP algorithms use the delay tolerance of applications to batch requests in time slots. Bandwidth assignment is made either at the end of the slot, or during the slot, on a ...

35 A hybrid handover protocol for local area wireless ATM networks
Chai-Keong Toh
December 1996 **Mobile Networks and Applications**, Volume 1 Issue 3

Additional Information: full citation, abstract, references, citings, index

Full text available: pdf(960.44 KB) terms

While handovers of voice calls in a wide area mobile environment are well understood, handovers of multi-media traffic in a local area mobile environment is still in its early stage of investigation. Unlike the public wireless networks, handovers for multi-media Wireless LANs (WLANs) have special requirements. In this paper, the problems and challenges faced in a multi-media WLAN environment are outlined and a multi-tier wireless cell clustering architecture is introduced. Design issues for ...

36 Fairness and load balancing: Coordinated load balancing, handoff/cell-site selection, and scheduling in multi-cell packet data systems
Aimin Sang, Xiaodong Wang, Mohammad Madihian, Richard D. Gitlin
September 2004 **Proceedings of the 10th annual international conference on Mobile computing and networking**
Full text available: pdf(312.56 KB) Additional Information: full citation, abstract, references, index terms

We investigate a wireless system of multiple cells, each having a downlink shared channel in support of high-speed packet data services. In practice, such a system consists of hierarchically organized entities including a central server, Base Stations (BSs), and Mobile Stations (MSs). Our goal is to improve global resource utilization and reduce regional congestion given asymmetric arrivals and departures of mobile users. For this purpose, we propose a scalable cross-layer framework to coordinat ...

Keywords: HDR, HSDPA, cell-site selection, handoff, load balancing, multi-cell, opportunistic scheduling

37 Estimation of reverse-link capacity for multiband DS-CDMA systems
Lei Zhuge, Victor O. K. Li
August 2000 **Proceedings of the 3rd ACM international workshop on Modeling, analysis and simulation of wireless and mobile systems**
Full text available: pdf(605.27 KB) Additional Information: full citation, abstract, references, index terms

In the future deployment of wideband DS-CDMA mobile communication systems, spectrum overlay among sub-bands with different bandwidths is probably inevitable. In this paper we present an approach to estimate the reverse-link capacity of overlaid multiband DS-CDMA systems in terms of the maximum number of users in each sub-band. We will derive the general capacity formula, and present a decomposition method for the capacity analysis to reduce the computational complexity. Based on thi ...

38 Bandwidth adaption algorithms with multi-objectives for adaptive multimedia services in wireless/mobile networks
Taekyoung Kwon, Ilkyu Park, Yanghee Choi, Sajal Das
August 1999 **Proceedings of the 2nd ACM international workshop on Wireless mobile multimedia**
Full text available: pdf(1.09 MB) Additional Information: full citation, references, citings, index terms

39 A utility-based power-control scheme in wireless cellular systems
Mingbo Xiao, Ness B. Shroff, Edwin K. P. Chong
April 2003 **IEEE/ACM Transactions on Networking (TON)**, Volume 11 Issue 2
Full text available: pdf(650.72 KB) Additional Information: full citation, abstract, references, citings, index terms

Distributed power-control algorithms for systems with hard signal-to-interference ratio (SIR) constraints may diverge when infeasibility arises. In this paper, we present a power-control

framework called utility-based power control (UBPC) by reformulating the problem using a softened SIR requirement (utility) and adding a penalty on power consumption (cost). Under this framework, the goal is to maximize the net utility, defined as utility minus cost. Although UBPC is still noncooperative and dis ...

Keywords: Nash equilibrium, Pareto optimal, admission control, cellular system, distributed algorithm, fairness, power control, robustness, signal-to-interference ratio (SIR), stability, utility function, wireless

40 Cluster resource management: Resource overbooking and application profiling in shared hosting platforms
Bhuvan Urgaonkar, Prashant Shenoy, Timothy Roscoe
December 2002 **ACM SIGOPS Operating Systems Review**, Volume 36 Issue SI

Full text available: pdf(2.00 MB) Additional Information: full citation, abstract, references, citings

In this paper, we present techniques for provisioning CPU and network resources in shared hosting platforms running potentially antagonistic third-party applications. The primary contribution of our work is to demonstrate the feasibility and benefits of overbooking resources in shared platforms, to maximize the platform yield: the revenue generated by the available resources. We do this by first deriving an accurate estimate of application resource needs by profiling applications on dedicated no ...

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IEE CNF	IEE Conference Proceeding
IEEE STD	IEEE Standard

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1. **Hybrid ARQ with selective combining for fading channels**
Qinqing Zhang; Kassam, S.A.;
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JOINT SOURCE CHANNEL CODING WITH HYBRID FEC/ARQ FOR BUFFER CONSTRAINED VIDEO TRANSMISSION

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Abstract - We propose an Automatic Repeat Request (ARQ) / Forward Error Correction (FEC) scheme for synchronous transmission of video over a binary symmetric constant rate channel. The approach consists of jointly allocating source and channel rates to video blocks from a given admissible set subject to the buffer or equivalently end-end delay constraints. The channel codes used are the popular class of powerful FEC codes known as Rate-Compatible Punctured Convolutional (RCPC) Codes. The method used involves independent coding of the video units and optimization of the end-to-end expected delivered video quality. The existence of a return channel is assumed through which the decoder informs the encoder about the success/failure of the transmission. In the event of a failure, incremental parity information is sent to the decoder for correcting errors and a reallocation performed at the encoder. The simulations done point out the efficacy of the proposed scheme.

INTRODUCTION

Variable bit rate methods for source coding of video have received significant attention in the past in view of their compression efficiency and network bandwidth utilization. The problem of optimal allocation of rates for delay constrained video has been solved for both the cases of noise free [1] and noisy channels [2]. In [2], the channel considered was a burst error wireless channel and the error control technique employed was conventional Automatic Repeat Request (ARQ) wherein based on feedback from the decoder, packets received in error are retransmitted.

In this paper, we employ a hybrid ARQ/FEC scheme using Rate-Compatible Punctured Convolutional (RCPC) codes [3] and investigate the potential advantages of integrating it into an ARQ based scheme to improve the delivered video quality at the receiver. RCPC codes are a family of efficient channel codes with rate compatibility restriction which implies that all the code bits of a high rate code are used by the lower rate codes. Such schemes have been considered for image transmission [4]. However there has been no treatment

in literature on the use of these codes in the context of delay constraints (which result in equivalent buffer constraints). Intuitively, a hybrid scheme based on RCPC codes has an edge over conventional ARQ in the sense that during retransmissions following an error, there is no need to resend the entire packet but only incremental parity information as proposed in [3]. Because of the rate compatibility of the channel codes, any parity sent previously (as a forward error correction component or as a retransmission), can be combined with the incremental parity to facilitate error removal at the cost of lesser bandwidth.

SYSTEM, PROTOCOL AND CHANNEL DESCRIPTION

The proposed video transmission scheme has applications in video on demand and other storage and video server applications. The system consists of an encoder (figure (1)) which also performs the role of a rate controller. The rate distortion characteristics of each of the video units are assumed available to the encoder. Based on this knowledge, the encoder allocates source rates and then channel rates to the video units. The transmission units are variable length packets. A typical packet consists of a source-channel code for a video unit and an outer error detection code such as a 16-bit CRC for the source code and finally a header.

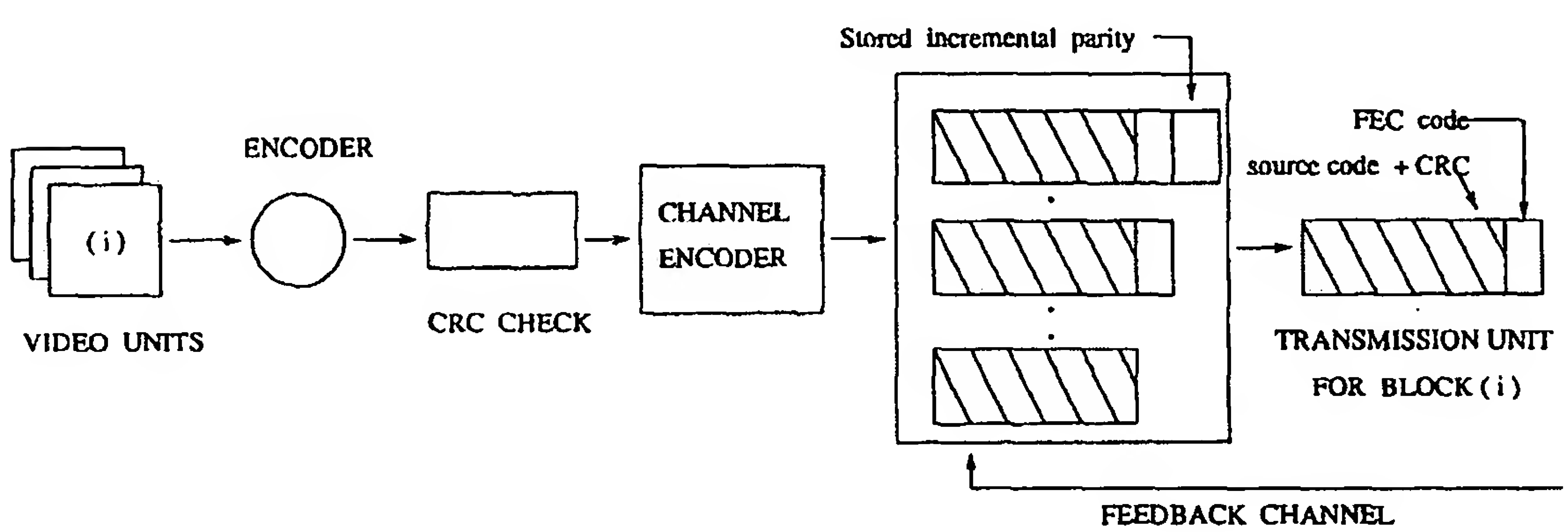

Figure 1: Encoder: Rate Control using hybrid FEC / ARQ

On receiving the bits, the decoder attempts to correct the channel errors and recover the packet. The success or failure in correcting the channel errors is determined by the error detection mechanism (assumed to be perfect) and the result is conveyed back to the encoder through the feedback channel. Upon success the encoder proceeds with the transmission of the next unit. Else, the encoder, optimally switches to a stronger channel code and transmits only the incremental bits needed for the chosen code. The decision to send more bits is taken only if the deadline for the display of the unit has not been reached and the procedure goes on.

The channel model used is that of a constant bit rate, stationary, binary

symmetric channel with instantaneous feedback. The rate allocation algorithm developed is particular to this simple model. However, it still highlights the salient features of the buffer constrained joint source channel optimization and can serve as a first step towards analysis of the problem using more complicated channel models. The framework can however, be easily extended to the case of feedback with delay.

PROBLEM FORMULATION

Definitions and Notations

Video Sequence: $(V_n)_{n=1,...N}$ is the sequence of transmission units to send. The units may be video frames, or subbands of a frame, or groups of frames; the only assumption being that each unit being sent is coded independently of others.

Source/Channel Rates: $S_k(i)$ and $C_j(i)$ denote respectively the source and channel rates in bits for the i^{th} unit of quality k and strength j respectively where $k = 1, ...K$ and $j = 1, ...J$. From the Rate-Distortion characteristic of the i^{th} unit, $S_k(i)$ is associated with mean squared Distortion $D_k(i)$. For convolutional channel codes, code strength j is associated with a rate factor $r(j)$ (so that $C_j(i) = r(j).S_k(i)$) and with a probability of bit error $p_b(j)$. The channel codes used are convolutional codes so that for $j_1 \geq j_2$, $r(j_1) \geq r(j_2)$ and $p_b(j_1) \leq p_b(j_2)$. The probability of bit error for a particular code strength depends only upon the signal to noise ratio in the channel. For our scheme we assume that the channel is stationary and the signal to noise ratio and hence $p_b(j)$ for each code strength j is known to the encoder.

Expected Distortion per Transmission Unit: Define probability of frame error $p_f(i)$ for the i^{th} unit as:

$$p_f^{k,j}(i) = 1 - (1 - p_b(j))^{S_k(i)}; \tag{1}$$

The Expected (Mean) Distortion for the i^{th} unit is defined as:

$$ED^{k,j}(i) = (1 - p_f^{k,j}(i)).D_k(i) + p_f^{k,j}(i).E(i) \tag{2}$$

where $E(i)$ denotes the energy of the i^{th} unit i.e. the mean square distortion encountered when zero rate is assigned to the i^{th} unit. This is clearly a pessimistic view since here a unit received in error is considered as lost fully.

Rate Constraints: Time is measured in terms of video unit times. Let N be the total number of units to be sent. Let

$$R^{k,j}(i) = S_k(i) + C_j(i) \tag{3}$$

denote the rate assigned to the i^{th} unit. Then if C denotes the channel bandwidth in bits per video unit time and L is acceptable latency of the application then at time n if unit m is currently being transmitted and $R'(m)$ bits of it are present in the encoder buffer we need:

$$R'(m) + \sum_{l=m+1}^{i} R^{k,j}(l) \leq (L+i-n).C \qquad for \;\; i = m+1, ...N \quad (4)$$

Statement of the Problem

Formulation (For the case of instantaneous feedback): Let the current time be n and the unit being transmitted be m. Let $R'(m)$ bits of unit m be in the encoder buffer. Now if the m^{th} unit is lost before time $n+1$, then the optimization problem can be formulated as:

For all $i \in m+1, ...N$ obtain $k(i)$ and $j(i)$ such that $k(i) \in 1, ...K$ and $j(i) \in 1, ...J$ and that

$$\sum_{i=m+1}^{N} ED^{k(i),j(i)}(i) + ED_{new}^{k(m),j_{higher}}(m) \quad (5)$$

is minimized subject to the constraints (4) where $R'(m)$ in (4) is replaced by $R'(m) + I(j_{low}, j_{higher})$ where $j_{higher} \in j_{low}, ...J$. The latter term is the incremental parity term for the m^{th} unit. For the case of delayed feedback, additional terms appear in (4).

Every time an error occurs, the encoder tries to send an incremental channel code for the unit in error so that (5) is minimized. So there is a dynamic reallocation for all the units which are yet to be transmitted. Since all the units up to n (current time) will already have been encoded, and there is a possibility that during the reallocation lesser number of bits are made available for them than the current assignment, we see that implementation is most elegantly supported if all the units are source coded in an embedded form. In this case taking away bits is equivalent to "chopping" off some bits off the source code.

Optimization Framework

The solution to the above problem is found using Dynamic Programming. In the trellis generated the stages represent the time in terms of video unit times and the states are the encoder buffer occupancy states (B_e) (related at successive time instants by: $B_e(i+1) = B_e(i) + R(i+1) - C$) as shown in figure 2. The constraints in (4) are implemented by restricting the number of states in each stage to $L.C$. The cost associated with a branch from node a in stage n to a node b in stage $n+1$ is given by the expected distortion associated with the rate $r(n+1)$ for the $(n+1)th$ unit $(r(n+1) = s(n+1) + c(n+1))$ where $r(n+1) = b + C - a$. Because of independent coding of the video units and the fact that the channel is a binary symmetric channel (memoryless) the cost of a branch is independent of the path it came from and hence we can use dynamic programming techniques. The metric in (5) represents the

total cost of a path from the initial to the final stage and is to be minimized over all possible paths.

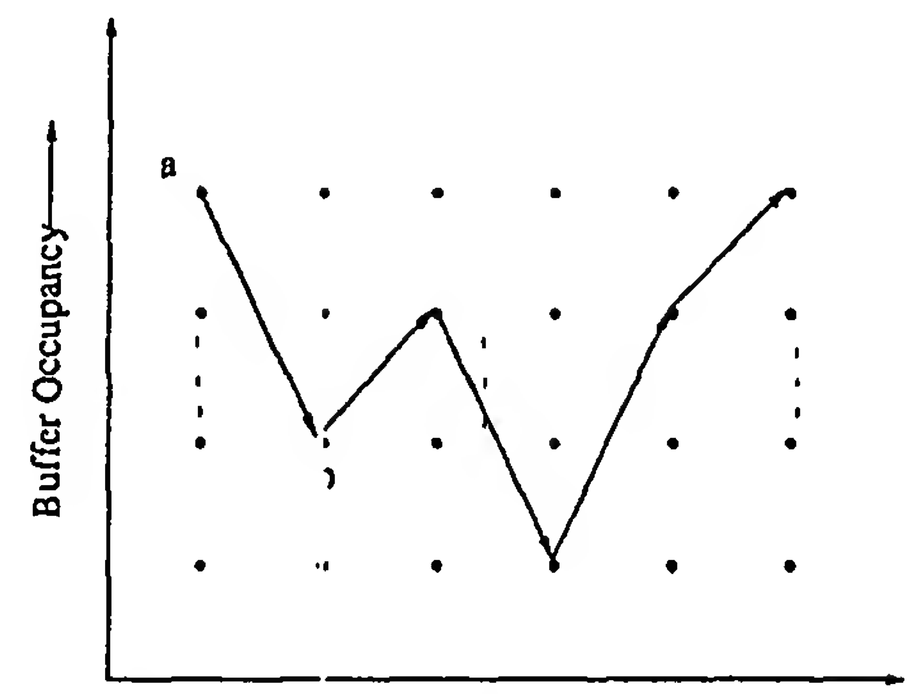

Figure 2: Trellis representing encoder buffer occupancy at different stages.

EXPERIMENTAL RESULTS

For the simulations, Said and Pearlman's (embedded) SPIHT coder [5] was used in the intraframe mode and the family of RCPC codes used (memory M=4 and puncturing period P=8) was obtained from [3]. The rate distortion characteristics used were a parameterized version of those for the "Football sequence" when coded as a sequence of images using the SPIHT coder. The simulations were run for 100 frames with end-end latency L=1. The value of L chosen was motivated by a desire to obtain representative performance for practical sequences (which typically have much larger lengths so that the end-end latency would typically be a small percentage of the playing time).

The video units (obtained after subdividing the output frames from the video player) and their sizes were chosen so that after source coding the video units the average transmission packet size was close to that needed for maximum throughput for the ARQ scheme for each value to signal to noise ratio in the channel. This would offer a conservative comparison. The hybrid scheme consistently outperforms the ARQ scheme for all ranges of bit error rates (figure (3)) with a striking difference for bit errors in the range (10^{-3} to 10^{-5}). A reason for large gains at high BERs is that because of the strict buffer size used (L=1), the ARQ scheme at high BERs is unable to get much throughput across.

CONCLUSIONS AND FUTURE WORK

We formalized the problem of optimal joint source-channel rate control for stored video and presented an algorithm to compute the optimal control which can be trivially extended to real time transmission. We considered

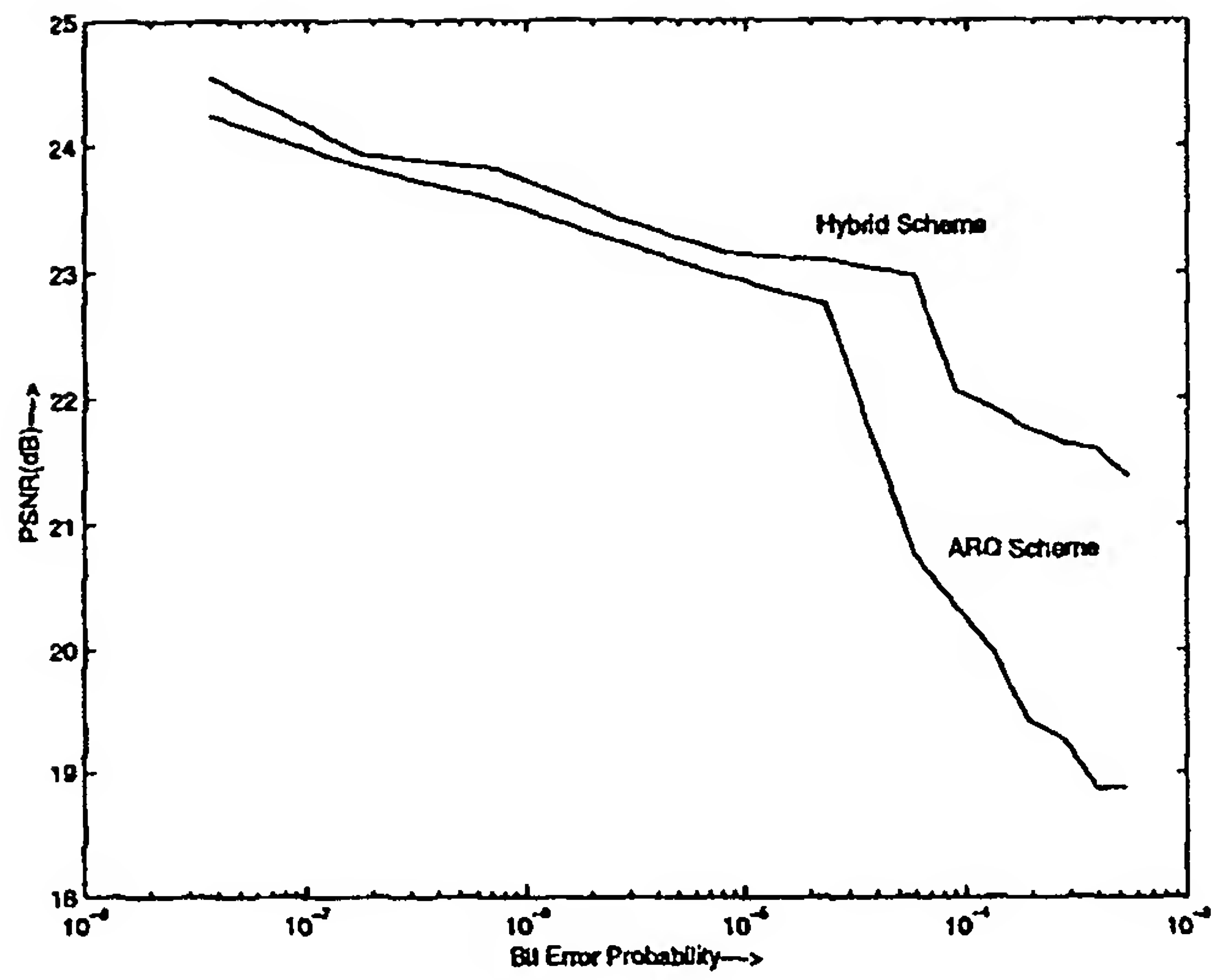

Figure 3: Average PSNR at the decoder as a function of signal to noise ratio in the channel for the hybrid and the ARQ scheme at a target transmission rate of about 1bpp.

the scenario where the encoder has knowledge of the channel state (signal to noise ratio) and the channel is stationary. It is to be noted that the scheme is optimal even for the case of time varying channel when the state is known to the encoder. Thus an extension of this work would be to incorporate a channel state estimator whose output is available to the encoder.

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A CDMA-based radio interface for third generation mobile systems

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This paper deals with the use of a CDMA-based radio interface in third generation mobile systems (Universal Mobile Telecommunications System – UMTS, and Future Public Land Mobile Telecommunications System – FPLMTS). The paper is not intended as a detailed analysis of the radio interface performance, but as an overview of the main issues arising in a typical CDMA-based mobile system, discussing the different available technical solutions. First of all, the basic requirements of the radio interface in a third generation mobile system are outlined. In particular, the support of variable bit rate transmission, the adaptability to the different propagation and service environments and the flexibility are felt to be important topics to be discussed. Then, the main characteristics of the CDMA access technique are depicted, in relation with the above mentioned requirements, focusing in particular on the DS-CDMA radio interface designed within the RACE II – CODIT Project. In that context the paper describes some of the technical solutions proposed for the provision of advanced features such as macrodiversity, multibearer transmission and variable bit rate services.

1. Introduction

First generation (based on analog technology) and second generation (based on digital technology) mobile communication systems consist of a set of different systems designed for specific purposes: paging, cordless, cellular mobile, satellite mobile.

At the beginning of the next century, a more advanced third generation of mobile communication systems is expected to be ready for the operational phase: the so called (according to the European terminology) Universal Mobile Telecommunications System (UMTS) or, referring to the ITU terminology, Future Public Land Mobile Telecommunication Systems (FPLMTS). For simplicity, in the following we will refer always to UMTS. UMTS will integrate, in a unique structure, all the services mentioned above that are currently provided by means of several ad-hoc systems. At the same time, it will make available anywhere and at anytime a wide range of services with both QoS and maximum bit rates higher than those provided by first and second generation systems. In particular, UMTS will support data services up to 2 Mbit/s.

This paper is focused on the analysis of the adoption of a CDMA radio interface for the support of UMTS services. The content of this paper is based on the experience of the authors within the RACE II – R2020 CODIT (COde DIvision Testbed) Project, partially funded by the European Commission. The paper is organised in the following way. First of all, the basic requirements of the radio interface in a third generation mobile system are outlined. Then, the main characteristics of the CDMA access technique are depicted, in relation with the above mentioned requirements. Finally, the radio protocols architecture of a CDMA-based system is presented: for each component, the possible technical solutions are shown, highlighting advantages and disadvantages and their impact on the overall performance and complexity.

2. Radio interface requirements

An advanced "third generation" mobile communication system must be able not only to replace all the mobile systems currently in operation, but it should also be extremely versatile and adaptable in order to face present and future unpredictable service and traffic demands. As a consequence, such a system should be based on an air interface satisfying a wide set of requirements that are briefly summarised in the following.

High adaptability

The radio interface should be able to easily adapt its characteristics in order to support any kind of service, both in terms of quality (BER, delay) and bit rate. Both single and multibearer capabilities should be supported so as to handle in a highly efficient way any kind of service request.

Low sensitivity to environment variations

The radio interface performance is expected to be nearly environment independent. This "robustness" with respect to environment and propagation variations can be obtained by means of a real time dynamic adaptation of the main radio transmission characteristics (e.g., modulation, coding, power control rate, etc.). Advanced system level solutions, such as macrodiversity, adaptive antennas and multilayer radiocoverage also contribute to meet this requirement.

Transport up to 2 Mbit/s (continuous and packetized)

A radio interface candidate for UMTS should support, theoretically, any bit rate in the range 0–2 Mbit/s, allowing real time variation during the call phase according to the source and system needs. With such a wide variation range, some kind of "intelligent device" responsible for resource allocation (number of time slots, bandwidth) to each call is mandatory so as to maximise the spectral efficiency. On a call basis, the radio interface should be able to satisfy the requests sent by the resource manager. Support of

packet transmission is also a mandatory feature: the radio interface should keep the radio resources busy only during the transmission of the packet; the resources have to be released (or, at least a significant part of them) as soon as the packet transfer is completed. This feature, together with the provision of asymmetrical transmission, is a key factor in the support of multimedia services.

Dynamic channel allocation

Dynamic channel allocation techniques are needed in order to optimise the spectral efficiency. An advanced radio interface has to be flexible enough to allow for the implementation of several channel allocation policies (e.g., centralised/distributed) that can vary according to the environment.

Optimisation of spectral efficiency

The expected growth of mobile communications, compared with the limited available spectral resources, will demand a frequency re-use factor approaching one (at least in the same coverage level). In addition, the presence of different kind of users (e.g., vehicular, pedestrian), with completely different mobility characteristics, will call for a multiple-level radio coverage, organised in a hierarchical structure. This scenario will also loosen the power control constraints, so that the related mechanism will more easily meet the interference minimisation requirements.

The overall process of the spectral efficiency optimisation will have to cope with the possible presence of multiple operators in the same area, each of them competing with each other and operating the related networks in an unregulated way.

Seamless handover and macrodiversity

The radio interface must provide seamless handover in any operational condition and should be able to support macrodiversity. Macrodiversity is defined as the process allowing the MS to be simultaneously connected to more than one base station. Macrodiversity can improve both QoS and robustness against shadowing.

Security

Due to the nature of the medium (i.e., a radio channel) used in mobile communications, particular attention should be paid to the provision of security features, covering the aspects of both user data confidentiality and user authentication.

High traffic density, coping with high peaks in hot spots

The selected radio interface should be able to handle very high traffic densities (e.g., in metropolitan or indoor environments), mainly concentrated in hot spot areas embedded in a low traffic density environment.

Simple network deployment

The expected unpredictable growth of both the number of users and the provided services requires a high degree of reconfigurability and an easy expansion of the mobile networks. Therefore the radio interface characteristics should ease the planning activities carried out by the operators in a rapidly evolving environment.

3. CDMA characteristics

CDMA (Code Division Multiple Access) is a particular kind of the so called Spread Spectrum communication techniques: this is a particular class of communication systems which make use of a bandwidth much greater than the minimum bandwidth required for the information signal to be transmitted [1,2]. The signal bandwidth spreading is obtained by processing the information signal by means of Pseudo Noise sequences (PN-sequences) having particularly good autocorrelation and cross correlation properties.

In a Direct Sequence CDMA (DS-CDMA) system all the users share contemporaneously the overall bandwidth W. The signal spreading is performed by multiplying the digital information signal (with bit rate R_b) by a PN sequence with bit rate (referred to as "chip rate") $R_c \gg R_b$. The desired information can be recovered by multiplying the overall received signal (superposition of all the user signals) by the particular PN sequence used at the transmitting side for that specific user. When conventional threshold detectors are used, in order to extract the desired information, it is extremely important that all the user signals are received at the BS receiver with the same power level. Note that this constraint can be loosen in case of adoption of multi-user receivers. The basic radio transmission chain of a CDMA system is shown in figure 1.

The behaviour of a CDMA system and its very strict power control requirement can be well understood by means of the so called "United Nations cocktail party" analogy. At this party, a lot of people located in the same hall (i.e., the same frequency band) are talking at the same time and in different languages (PN-codes). Everybody can "select" and understand a conversation from the set if the spoken language (i.e., the code) is known. This is feasible until the number of the simultaneous conversations becomes so high that the overall background noise does not allow the people to understand each other, even if the language is known.

Moreover, if during the party two persons are talking to each other very loudly (i.e., using a high power level), they can understand each other very well but, at the same time, they make the communication between all the other people impossible. This example shows the importance of the power control mechanisms in CDMA mobile systems: efficient algorithms have to be implemented in order to cope

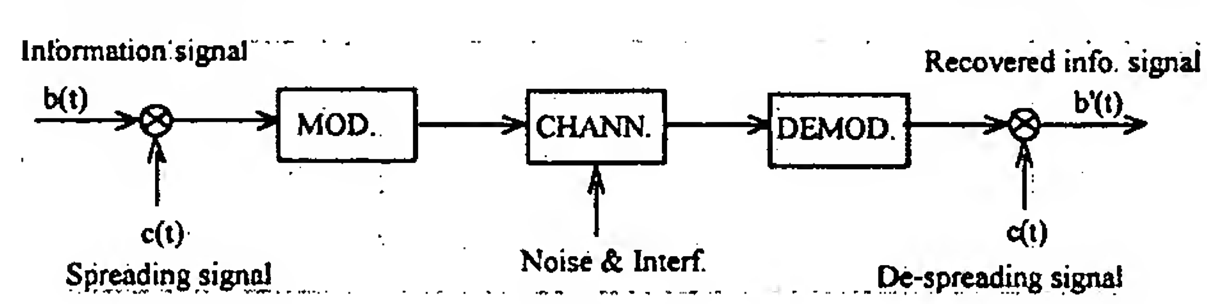

Figure 1. Basic CDMA Tx–Rx chain.

with the so called "*near far effect*" (i.e., signals coming from mobiles *near* the base station must not overwhelm signals coming from mobiles *far* from the base station). The previous analogy clarifies immediately that CDMA systems are interference limited: the number of users can increase (without any hard limitation) as long as the overall interference level on the air allows to meet a given QoS requirement.

Considering the uplink of an isolated perfectly power controlled cell, the previous considerations can be put in a formula giving the well known expression of the number of users a cell can support with a given quality of service [2]:

$$C = \frac{g}{E_b/N_0},$$

where $g = R_c/R_b$ is the processing gain and E_b/N_0 is the operating point for the considered service; the users are assumed continuously active and the base station equipped with an omni-directional antenna.

From figure 1 it is possible to see one of the main characteristics of CDMA: the capability of exploiting the activity of the source (Service Activity Factor – SAF). Usually the information sources are not continuously active: they are generating information only for a percentage SAF of the time. For example, referring to the speech service, experimental measurements show that voice is active about 38% of the time [3]. During silences the user signal is "zero" and hence, no power is radiated (i.e., no interference is created). This turns automatically into an average capacity increase of a factor of 1/(SAF). Of course, this is only a rough estimate of the capacity improvement due to discontinuous transmission. A more detailed analysis which is not based only on average interference considerations can be found in [4]. In practical systems, this capacity gain is lower because the bit rate cannot be reduced to zero (a certain minimum amount of information has to be transmitted for synchronisation or power control purposes). The cell capacity can be further increased by means of sectorization. In fact, if the cell area is covered by G_a sectorial antennas, the interference level detected by each antenna is $1/G_a$, which is lower than that in the omnidirectional case (i.e., the capacity is G_a times higher). Summarizing, the cell capacity is given by

$$C = \frac{gG_a}{(\mathrm{SAF})E_b/N_0}.$$

In case we are not considering an isolated cell but a multicellular system, the cell capacity will be reduced by a factor, which takes into account the interference coming from the surrounding cells.

The intrinsic characteristics of CDMA (possibility of simultaneous transmission of many users in the same RF band, exploitation of multipath, high immunity against narrowband interference, easy provision of multiple bit rate), meet already some of the requirements listed in the previous section.

3.1. CODIT system

Within the RACE Project CODIT [5,6], the detailed study of both a testbed and a system concept based on a DS-CDMA radio interface has been performed, trying to satisfy all the requirements of an advanced mobile communication system. The key issue in the design of CODIT was "flexibility". In order to obtain an open multi rate radio interface able to support the very wide range of UMTS services, three different chip rates have been foreseen: 1.023 Mchip/s, 5.115 Mchip/s and 20.46 Mchip/s whose correspondent RF bandwidths are about 1 MHz (narrowband channel), 5 MHz (mediumband channel) and 20 MHz (wideband channel). During the call set up phase, as soon as the requested service is known, a resource manager performs the mapping of the information bit rate onto the most appropriate chip rate so as to satisfy the QoS requirements. During this step, besides the spreading bandwidth choice, also the channel coding and interleaving schemes are selected by a specialised entity called "configuration unit". One of the most important parameters during this phase is the maximum information bit rate R_{bMAX}: the chosen spreading bandwidth must guarantee that the minimum processing gain $g_{min} = R_c/R_{bMAX}$ does not drop below a certain value.

Variable bit rate

In CODIT, all the physical channels are organised in time periods (called "CDMA frames") of 10 ms. Every 10 ms, depending on the source needs, a different number of information bits are transmitted over a frame of the so-called PDCH (Physical Data CHannel). The PDCH is the physical channel obtained by the spreading of the DICH (Dedicated Information CHannel), which is the logical channel resulting from the multiplexing in time of the Traffic Channel (TCH), carrying the user traffic, and the Dedicated Control Channel (DCCH), carrying the related signalling traffic. In parallel to each PDCH, a Physical Control Channel (PCCH) is continuously transmitted so as to provide the receiver end with the fundamental information of the current bit rate (i.e., the number of bits that have been carried by the current CDMA frame). For example, referring to the speech service, the codec designed within the project can have 7 different (net) output rates, from a minimum of 0.4 kbit/s to a maximum of 16 kbit/s. The relative transmitted power during a 10 ms frame is a function of the bit rate: the higher the bit rate, the higher the transmitted power. In this way, there is no waste of resources: when a short voice segment has low information content, it can be encoded with fewer bits that will be transmitted at a lower power level, thus, minimising the interference toward the other users. Figure 2 shows an example of the relative transmitted powers between PCCH (fixed bit rate 4 kbit/s) and PDCH (variable bit rate).

Power control

In order to guarantee that at each cell site the mobile user signals are received at the desired level, both open and

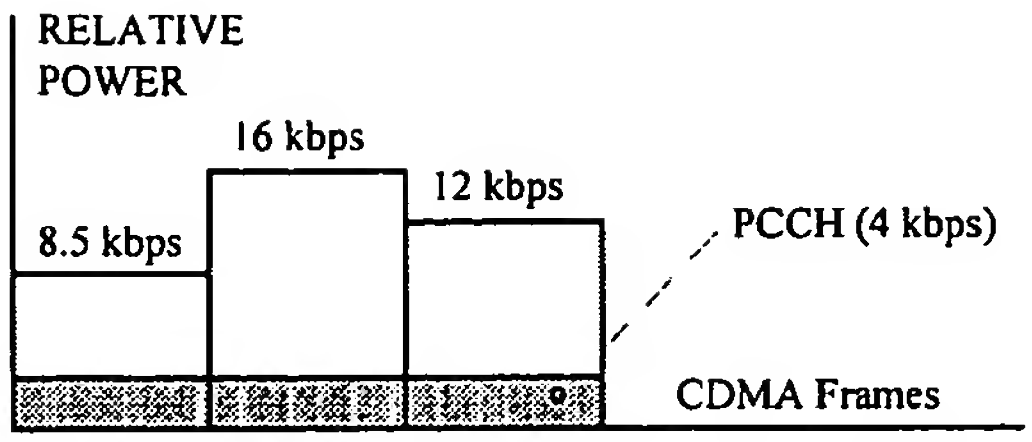

Figure 2. Relative transmitted powers in case of variable bit rate.

closed power control loops have been adopted. The open loop, based on the signal received from the pilot channel (a known beacon channel) transmitted by each BS, copes with the signal variations due to shadowing, while the closed loop adjusts the signal level so as to compensate the variations due to fast fading. At the base station, the received power from each PCCH (on the uplink) is continuously compared with the desired threshold and then, appropriate correction commands are sent to the mobiles through the corresponding PCCHs on the downlink. In CODIT, the power control information is transmitted over the PCCH every 0.5 ms unprotected (i.e., the power control bits are not encoded).

Cellular coverage, handover and macrodiversity

In CDMA systems, different signals are distinguished between each other by means of the spreading codes (all the users are sharing the same RF band at the same time, i.e., the reuse factor is equal to 1). This implies that CDMA cellular systems require no frequency planning. This is really one of the strong points of CDMA and it is extremely important in case of future communication systems like UMTS where, at the moment, the users demand cannot be exactly estimated. If the traffic load is increasing more than expected, it is sufficient to add a new BS, without performing any frequency re-planning like in TDMA systems.

The other important aspect related to the unique RF channel is the easy implementation of handover and macrodiversity: during the movement between cells, the communication is handed over from one BS to the other, at most changing a PN-code (no change of frequency is required). This is the so-called "soft-handover". At the same time, the universal frequency re-use allows the simultaneous reception of the mobile signal by more than one base station (macrodiversity reception). Macrodiversity is the situation where a MS is connected to more than one BS at the same time. This technique allows to reduce fading and shadowing effects, automatically providing a seamless handover. The MS monitors continuously the power level received from the pilot channels of the surrounding BSs: when this is above a given threshold, the corresponding BS enters the "active set" (i.e., the set of the BS that are communicating with the same mobile in macrodiversity mode). In a similar way, the BSs "badly" received are removed from the active set. An "ad hoc" entity on the fixed network side will perform some kind of selection/combining of the signals originated by the same mobile and received through different BSs (see section 4.2.1).

On the downlink each BS belonging to the active set transmits the same information toward the MS: at the MS the rake receiver [1,2] will exploit the path diversity performing the maximum ratio combining of the best rays received through different paths. In other words, the rake receiver is able to perform the selection and the coherent (vectorial) weighted sum of the best rays (coming from the same or different BSs) associated to the same information signal.

Even if CDMA allows the use of a unique RF channel, the radio coverage in CODIT is provided by means of a hierarchical cellular structure of picocells, microcells and macrocells, where each cell layer is operating in different RF bands. This choice makes the system more robust against imperfection of the power control algorithms in neighbouring cells with different size. Of course in this case the soft handover can be performed only within the same hierarchical level. When (for mobility issues or for signal weakness) a handover between different hierarchical levels is needed, this will require a change of frequency (inter-frequency handover). A specialised procedure called "compress mode" allows to perform a seamless inter-frequency handover. When an inter-frequency handover is required, the CDMA frame is split into two half-frames: the first half-frame is used for the communication with the current RF channel, while the second half-frame is used for the communication with the new cell (operating on a different RF carrier). As soon as the connection with the new target cell is stable, the compress mode is released and the standard transmission mode starts again with the new cell.

An important remark about synchronisation during macrodiversity and inter-frequency handover has to be introduced. CODIT is an unsynchronised system: the BSs are not synchronised. However, during the macrodiversity operation, the different radio links connecting MS and BSs have to be synchronised to each other. This is required, in order to perform the combining of the signals received at the MS from different BSs.

Signal spreading

Signal spreading in CDMA systems can be performed adopting two different main approaches: the synchronous spreading (based on short codes) and the asynchronous spreading (based on long PN sequences). The first solution allows the design of orthogonal code sets and, more generally, a control of the mutual interference; the drawback consists in the need of an accurate code management (the number of the short codes is limited), particularly in case of handovers. The use of long spreading codes, even if does not allow to get some of the benefits provided by the orthogonal codes (e.g., the very high reduction of intra-cell interference in the downlink), has some relevant practical advantages. In fact, the number of long PN sequences is virtually infinite, making then unnecessary any code management functionality (the probability that two users are

using the same code is negligible). Moreover, long PN sequences do not need any kind of synchronisation (neither between channels, nor between BSs) and present a good flexibility with respect to variable rate services. In CODIT, both short and long PN sequences have been adopted: short codes are used on the Pilot, Synchronisation and Random Access Channels while long codes are used for the remaining channels (PDCH, PCCH, Broadcast, Paging and Access Grant Channels). In particular, the long codes are obtained giving different phase shifts to a PN sequence with period $2^{41} - 1$.

An important characteristic of DS-CDMA systems is related to security. The spreading process performed by means of a PN sequence that is unknown to non authorised users provides automatically a certain level of confidentiality. The protection level is not very high, but it could be acceptable for all the users who do not require very secure exchange of information (that can be provided only by means of encryption).

4. System structure and protocol architecture

This section briefly presents the architectural assumptions taken in the development of the CODIT system and the structure of the specified radio protocols. The discussion then focuses on the implementation of specific topics related to the innovative features needed for the support of UMTS services.

4.1. Network reference model

In order to design the radio protocols and the system management functions for a CDMA-based UMTS system, the CODIT Project assumed the system architecture depicted in figure 3. This classical architecture includes four functional entities, namely the Mobile Station (MS), the Base Station (BS), the Radio Network Controller (RNC), and the Mobile Control Node (MCN).

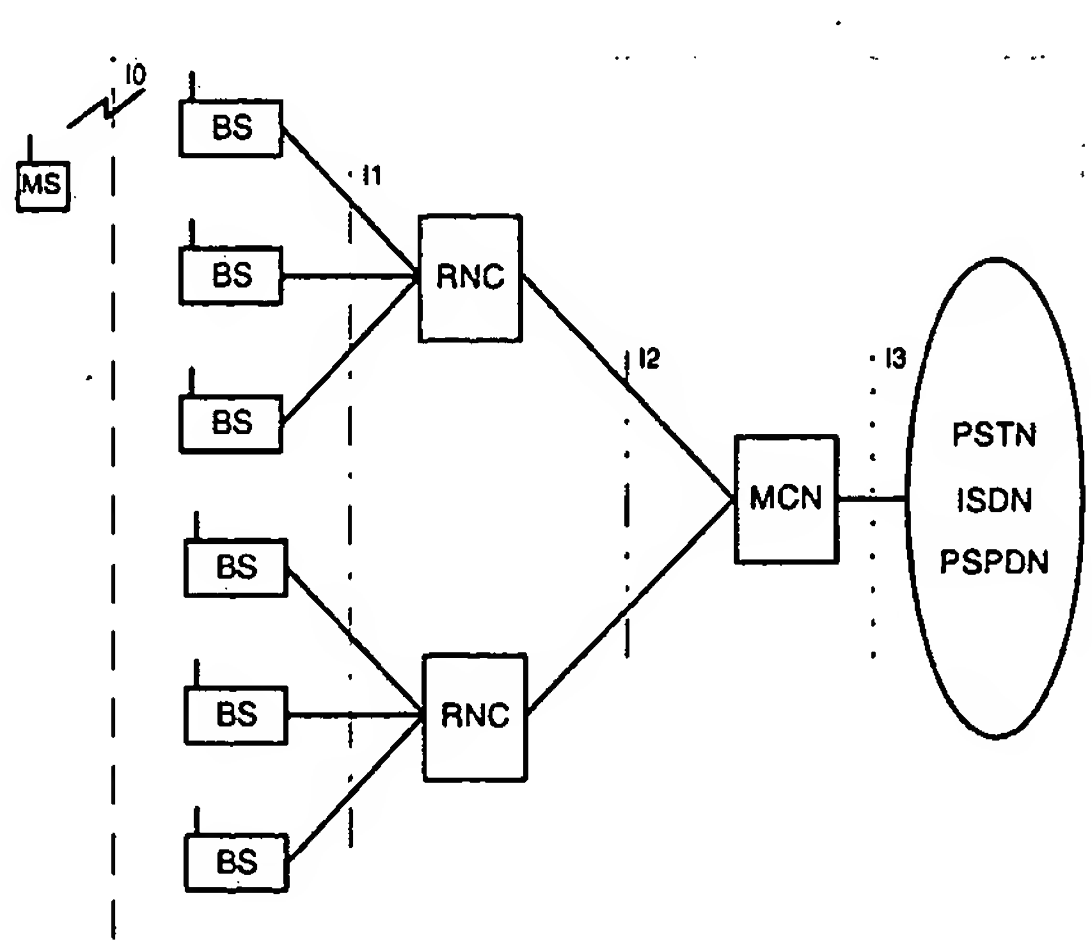

Figure 3. CODIT UMTS networks reference model.

The Radio Network Controller manages the macrodiversity functionality: the frame selector (choosing the best frame coming from the different BSs simultaneously connected to the same MS) is located within the RNC. In this case, the RNC identifies a macrogroup (i.e., the whole set of BSs that could be involved in a communication with the same MS).

The previous paragraph applies to the classical definition of macrodiversity, but in a CDMA system also adjacent macrogroups can use the same portion of the frequency band (a continuous coverage layer could use a single portion of the frequency band). Therefore, higher layer macrodiversity functionality is provided by the Mobile Control Node, choosing the best frame or message coming from the RNCs in order to allow macrodiversity also outside the classical macrogroups.

The Mobile Control Node can mainly be seen as the switching centre that allows the interconnection of the UMTS network to the fixed network (PSTN, ISDN or PSPDN). The MS can communicate with a single BS or with several BSs in case of performing handovers or macrodiversity reception. A number of BSs communicate with a single RNC through one interface. Similarly, a number of RNCs are connected to a single MCN, which in turn connects to the fixed network.

4.2. CODIT radio protocol architecture

The architecture of the radio protocols of the CODIT [7] system is layered in accordance to the Open Systems Interconnection (OSI) Reference Model developed by the International Standards Organisation (ISO). However, some deviations from the OSI Reference Model have been included in order to cope with some specific radio interface characteristics of a CDMA-based UMTS system.

A GSM- and a DECT-like protocol architecture have been studied for their applicability to define the CODIT UMTS radio protocols. Based on the investigation results, a DECT-like protocol architecture has been adopted, mainly because macrodiversity reception and multi-bearer connections can be implemented more easily. The chosen architecture type also provides high flexibility in allocating proto-

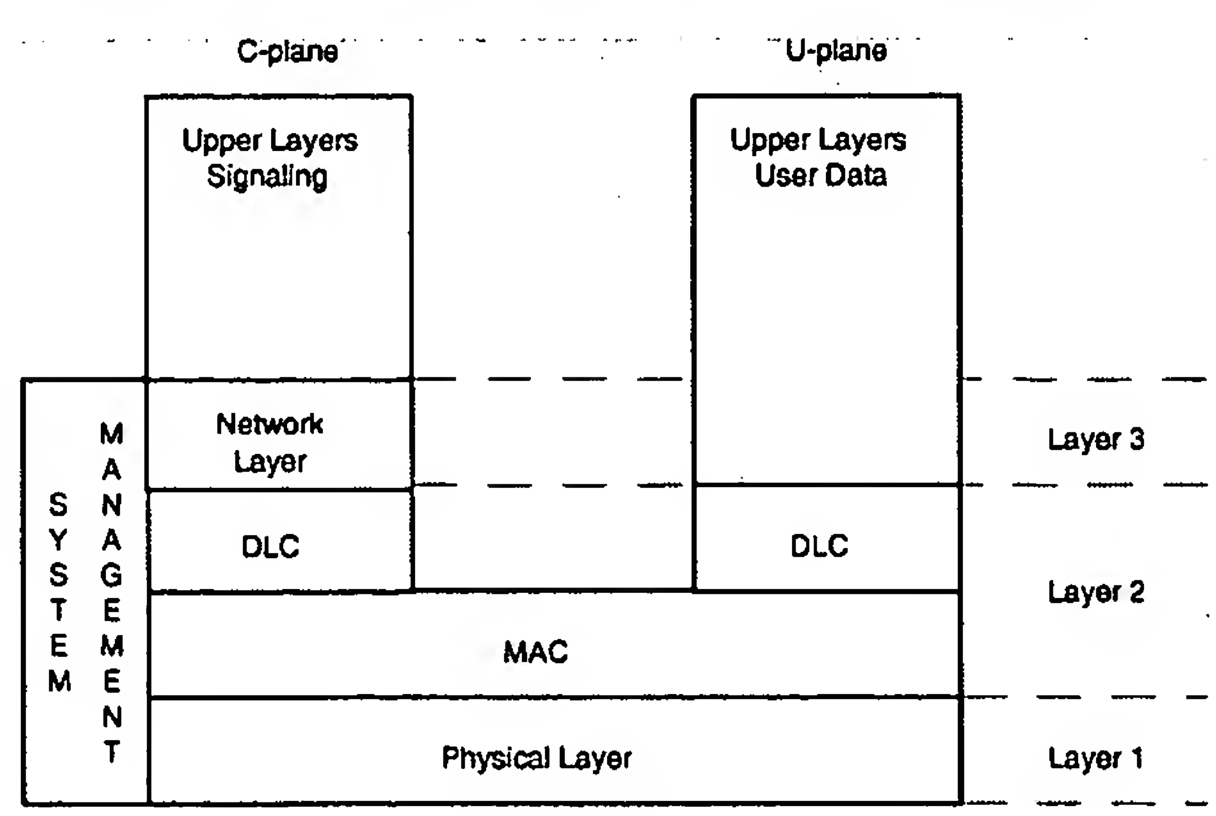

Figure 4. Overall protocol architecture.

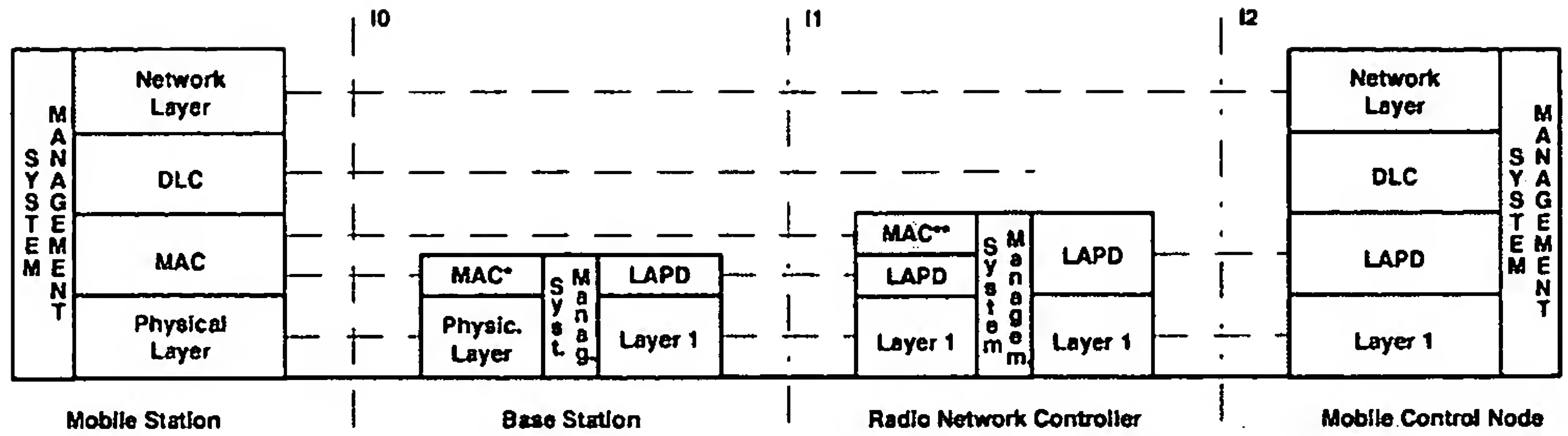

Figure 5. Protocol end-points.

col end-points to the network entities and in implementing handover procedures.

The CODIT radio protocol architecture (figure 4) includes two layered protocol planes and an unlayered system management entity for performing various link control, system maintenance and network tasks. The protocol stack is separated into a control (C-) and a user (U-) plane in order to process signalling and user data information separately in independent protocol entities.

The unlayered System Management Entity, which is a deviation from the OSI Reference Model, mainly includes all radio link control algorithms, and exchanges signals with the layered radio protocol stack in order to request the execution of some signalling procedures or to receive measurements from the protocol stack.

The protocol stack itself, namely the C-plane, provides the needed synchronisation between the C-plane and the U-plane for a specific network layer call. As a matter of fact, this synchronisation is needed during specific phases of the call, such as call establishment, handover and call release, when the U-plane data flow must be switched according to the C-plane based procedures for the connection management. Besides performing a synchronous switching of U- and C-planes for traffic data and signalling, the radio protocol stack must also rearrange the relationship of entities in both planes after the completion of the handover process. The synchronisation of the handover switching is also performed taking into account the service provided in the call, because different requirements about the used coding scheme and interleaving multiframes can arise.

In figure 5, the structure for the system interfaces (I0, I1 and I2) is shown including the end-points for the different protocol layers and sublayers. These interfaces can be considered as logical interfaces between functional entities and then, in the physical implementation, can be easily grouped in different manners, thanks to the modular and flexible nature of the CODIT radio protocols. It should be noted that figure 5 is intended to show the location of the CODIT protocol end-points. All the other protocols (e.g., LAPD) shown in the figure only represent a possible choice for the testbed implementation.

Two different handover procedures are provided in the radio protocol architecture:

- bearer handover: it is controlled by the Medium Access Control (MAC) layer and allows the MS to change the bearers used for the current call, while maintaining the same service provided to the Data Link Control (DLC) layer;
- connection handover: it is controlled by the DLC layer and allows the MS to change the MAC layer used for the current call, while maintaining the same service provided to the layer 3.

The presence of distinct handover and macrodiversity functionality at the MAC and DLC layers allows flexibility in allocating the protocol end-points in the UMTS system.

Referring to figure 5, the macrodiversity functionality is located in the upper part of the MAC sublayer and is thus based on MAC frames (the frame selection process chooses the best received MAC frame). This solution implies a satisfactory macrodiversity efficiency: as a matter of fact, the frame selection is performed at a low layer, where the shorter frames are used, and then the RNC can build up the DLC frame using the best MAC frames received on the different connections. Also DLC macrodiversity functionality is included in order to completely exploit the advantages provided by the CDMA access scheme.

Figure 6 better details the architecture of the MAC sublayer. It is split into a lower part (MAC*) and an upper part (MAC**). There is always one instance of MAC**, but multiple instances of MAC* may exist. In the fixed part of the radio network, according to the protocol end-point allocation of figure 5, the functionality of MAC** is

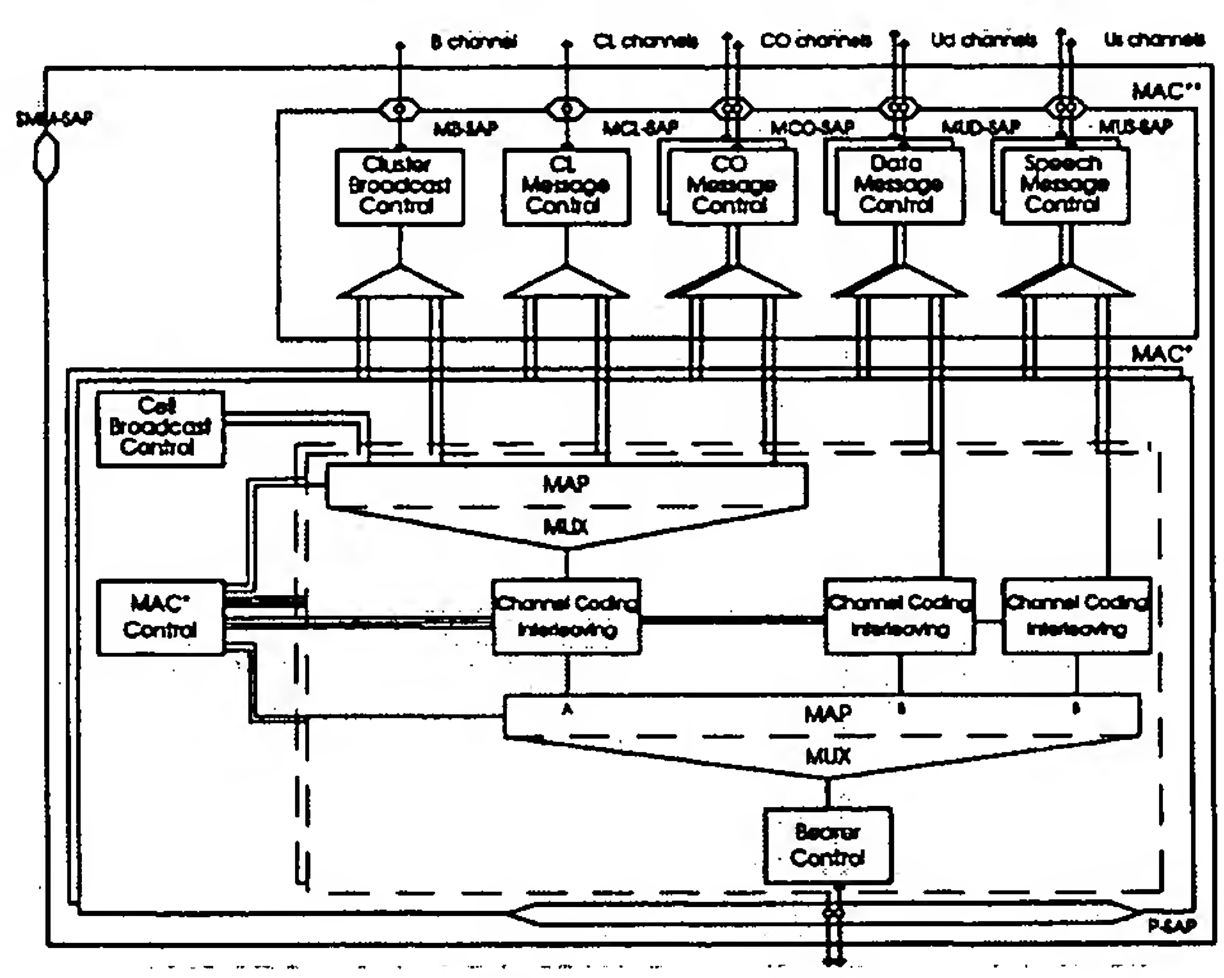

Figure 6. MAC sublayer architecture.

implemented in the RNC, while a single instance of MAC* is assigned to each BS connected to the RNC.

The lower part of the MAC sublayer (MAC*) mainly provides functions required to control the information exchange between a single BS and all MSs within a cell, to broadcast cell control information, to allocate and release physical channels established within the same cell, to provide functions for channel (de)coding and (de)interleaving, and to (de)multiplex information stemming from different logical channels of the C- and U-plane (from) onto physical channels provided by the physical layer. The upper part of the MAC sublayer (MAC**) mainly provides functions required to control the information exchange between the RNC and all MSs within the cluster of cells controlled by the RNC, to control multi-bearer connections, to provide random access capabilities to a MS, to broadcast cluster control information, to perform bearer handovers along with the required switching, and to perform macrodiversity reception in the RNC. A bearer handover occurs when an MS changes traffic channel and thus its physical layer within a BS or between different BSs connected to the same RNC. Separate protocol entities are provided in MAC** for controlling the transmission of broadcasts, the exchange of connectionless (CL) and connection-oriented (CO) information via the C-plane, and the exchange of user data (Ud) and speech messages (Us) via the U-plane.

After this brief overview of the CODIT radio protocol stack, the description will focus on three different control aspects that are felt to be fundamental in a UMTS system: macrodiversity, multibearer and variable bit rate control.

4.2.1. Macrodiversity control

As previously mentioned, in a CDMA system adjacent BSs can use the same portion of the radio spectrum and then it is possible to have a continuous radio coverage where the same RF channels are used by all the BSs. Such a scenario implicitly allows for the implementation of a macrodiversity functionality.

In principle both the MS, for the downlink macrodiversity, and the fixed part of the UMTS network, for the uplink macrodiversity, should use a frame selector entity in order to choose the best frame coming from the different radio connections. Actually on the downlink, where all the involved signals are received by the same equipment, macrodiversity can be simply resolved by the MS at layer 1 (see section 3), combining the received signals, thus improving the macrodiversity efficiency with respect to a higher layer frame selection process. On the uplink, the frame selection entity must be located on a higher hierarchical network level, with respect to the end point of the layer 1 (i.e., the BS), having to choose between frames coming from different BSs. This implies that the macrodiversity selection must be performed at a higher protocol layer (i.e., MAC sublayer) and cannot be directly executed on the layer 1 signals.

The structure of the CODIT radio protocols architecture and the allocation in the fixed network entities of the protocol end-points (see figure 5) imply that the macrodiversity frame selector functionality should be located in the MAC** sublayer within the Radio Network Controller (RNC). This solution provides a complete and efficient macrodiversity functionality to the MSs communicating with different BSs connected to the same RNC. However, from the radio interface point of view there would be no objections in allowing macrodiversity also between BSs connected to different RNCs, due to the previous mentioned continuous coverage using the same RF channels.

Providing the frame selector functionality in the RNC only, possible connections to BSs belonging to other RNC groups are useless, being impossible to combine the frame streams coming from the different RNCs. Therefore the CODIT system supports also the presence of a DLC frame selector functionality located in the Mobile Control Node (MCN). Possibly this selection should not use the soft information related to the radio channel, but it could simply be performed exploiting the DLC layer error protection mechanisms (e.g., the best CRC checking result). In such a way, the CODIT protocol architecture can also provide a higher layer macrodiversity functionality, even though in a less efficient way with respect to the MAC one, to the MSs communicating with BSs connected to different RNCs (within the same MCN group).

In figure 7, two examples of the macrodiversity scenarios supported by CODIT are presented. In this figure, MS#1 is communicating with two BSs connected to the same RNC, while MS#2 is communicating with two BSs connected with two different RNCs.

As previously mentioned, the protocol end-point allocation of figure 5 can be seen as an example without any constraints on possible changes in the network architecture. As a matter of fact, the idea of providing the macrodiversity and handover functionalities in the different protocol layers, and then allocating the protocol end-points to the network entities, allows for a complete flexibility in the UMTS system choices.

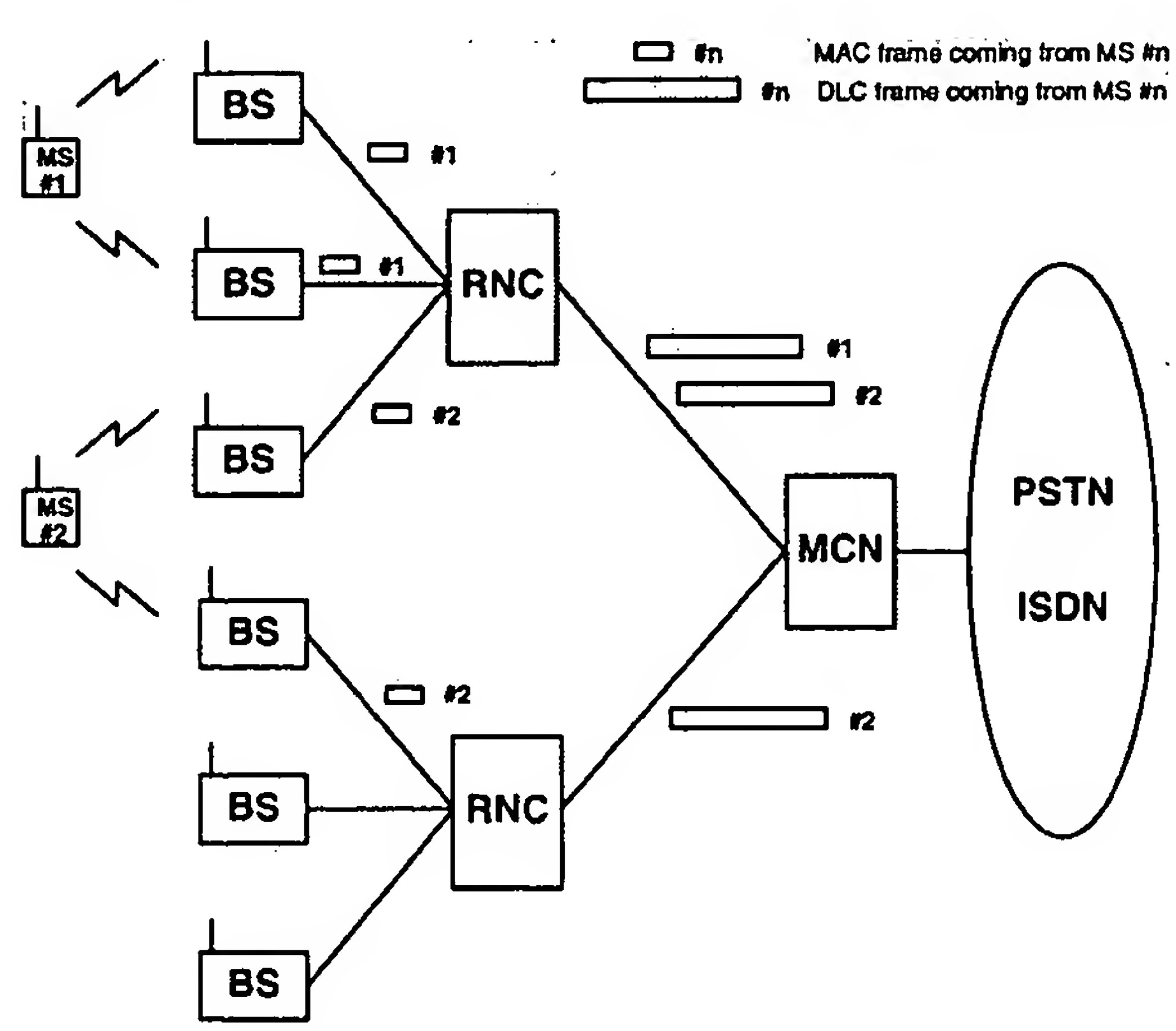

Figure 7. Macrodiversity scenarios.

4.2.2. Multibearer control

In CODIT system, multiple narrowband channels can be optionally combined to establish high bit rate services. The relevant splitting and combining functions can be either located at the physical layer or at layer 2 of the radio protocol stack. This section analyses the two options and explain the CODIT choice of including these functionality in layer 2.

In the following the terms "bearer" and "physical channel" can be considered as interchangeable, indicating a unique basic channel identified by one frequency, one spreading code and, if that is the case, one time-slot.

Multibearer splitting/combining at layer 1

The solution of allocating the splitting/combination of many physical channels in order to provide high bit rates at layer 1, together with the application of the Channel Coding functionality to the resulting logical channel, allows the spreading of errors coming from one RF channel over the whole logical channel. Of course an advantage in the error correction process can only be achieved, for data services requiring multibearer connections, if each bearer makes use of a different frequency (RF channel). As a matter of fact, in case of multibearer connections implemented using different spreading codes on the same RF channel, no advantage seems to arise from this solution with respect to perform the channel coding independently over each single bearer.

Moreover, many disadvantages come out from the decision to handle multibearer connections at layer 1. From the overall system point of view, the most critical issues are those related to the efficiency and flexibility of the macrodiversity functionality.

Since the layer 1 protocol end-point is located, on the network side, in the Base Station (BS), this solution implies that all the bearers, forming a multibearer connection, must be established with the same BS. This constraint reduces the system flexibility due to the fact that each single bearer cannot be handled independently: e.g., all the bearers forming a multibearer connection must be handed over simultaneously to a new BS. The need of handing over simultaneously all the physical channels supporting a multibearer connection can increase the probability of call dropping. In fact, if a multibearer connection consists of a certain number n of different bearers, the new BS must have n physical channels available, otherwise the handover procedure cannot be completed successfully.

The up-link macrodiversity functionality is performed in the RNC (Radio Network Controller) choosing the best frame coming from different BSs. The control of multibearer connections at layer 1 implies a less efficient macrodiversity algorithm, the choice of the best frame being performed on the whole logical channel instead of each single physical channel (see figure 8). In order to accomplish the spreading of transmission errors over the different RF channels, it is mandatory to handle large logical channel frames, encoding and spreading them over shorter physical channel frames. In this way the RNC frame selector can only deal

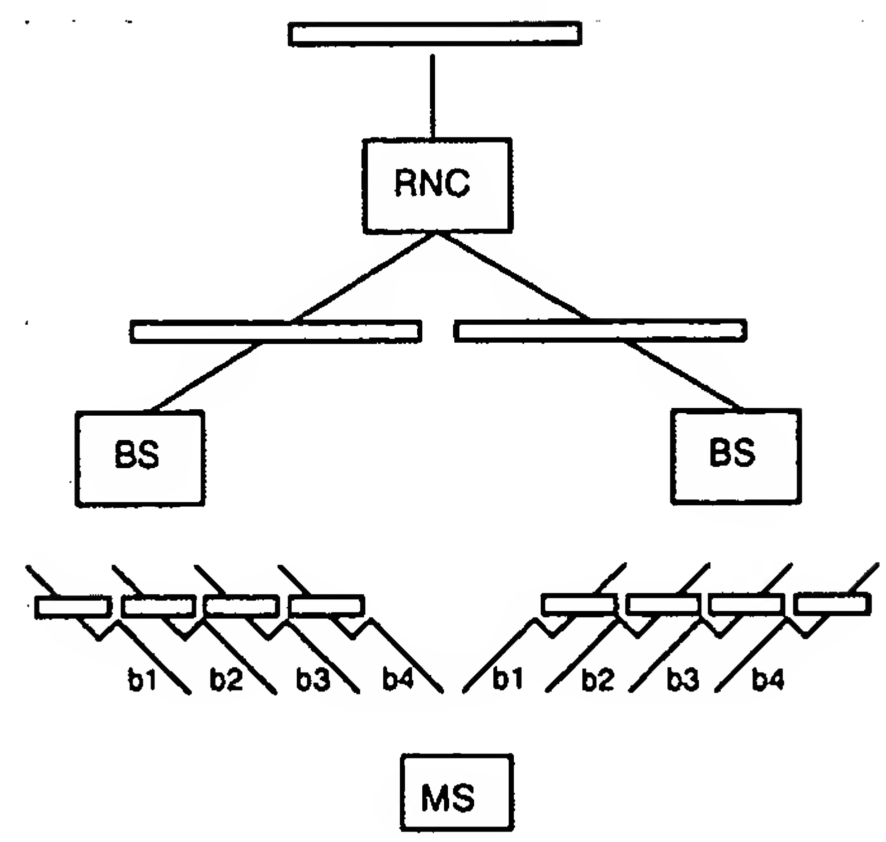

Figure 8. Macrodiversity in case of multibearer control at layer 1.

with logical channel frames, because layer 1 is terminated at the BS site and the channel coding is performed on the logical channel (and then the quality indicators, such as CRC and FEC fields, are present only at that level). Moreover, in order to allow the up-link macrodiversity, the same set of frequencies must be used by all the involved BSs: indeed only a complete duplication of the whole logical channel (i.e., the complete set of physical channels) is allowed. A scenario where only a part of the physical channels used in the multibearer connection are in the macrodiversity state cannot be achieved using this solution.

Another drawback of this solution can be found in case of the loss of one bearer forming the multibearer connection: the whole logical channel must be released because the channel coding entity needs all the bearers in order to perform its task. The requested service cannot be provided anymore and a change in the data rate seems to be very complicated. As a matter of fact, in case of offering, for a while, a lower data rate service, the whole channel coding scheme should be re-negotiated giving many problems of synchronisation between the two ends and introducing delays.

Due to the fact that for each data rate service a different channel coding is used, it seems also very difficult to implement efficient variable data rate services, for the same reasons explained in the previous paragraph.

Multibearer splitting/combining at layer 2

With reference to the CODIT radio protocol architecture, the multibearer control entity seems to be better located in the upper part (MAC**) of the MAC (Medium Access Control) Sublayer, within layer 2 (see figure 5). Being the end-point of this sublayer located in the RNC, this solution allows to avoid the constraints previously highlighted.

All the physical channels, forming the same multibearer connection, can be handled as completely independent carriers, i.e., each bearer can be established with a different BS and can be independently handed over. This feature allows a better and more efficient radio resource management within the set of BSs controlled by the same RNC, when high data rate services are requested. Moreover, the up-link

macrodiversity becomes more efficient allowing the frame selector, located in the RNC, to choose the best frames coming from the involved BSs on a single physical channel basis. In this way, also the flexibility of the macrodiversity scheme is improved due to the possibility of implementing a kind of "asymmetrical macrodiversity", where only a subset of the used physical channels is in the macrodiversity mode (see figure 9).

The scenario presented in figure 9 could be supported by the MAC protocol functional entities presented in figure 10. In that figure, all the entities included in the MAC** sublayer (within the dashed rectangle) are located in the RNC, while the MAC* and the layer 1 are terminated at the BS site.

The previous scenario and the improvement of the macrodiversity functionality can only be achieved if the channel coding is performed within the BS at MAC* level on a single physical channel basis, because of the required independent control of the different bearers involved in the same multibearer connection. This solution also allows a better traffic load balance among adjacent BSs: the possibility of establishing independent bearers with different BSs allows a better distribution of the user traffic load between them. As a matter of fact the minimum allocable resource is now the single bearer and not the whole logical channel as in the physical layer solution.

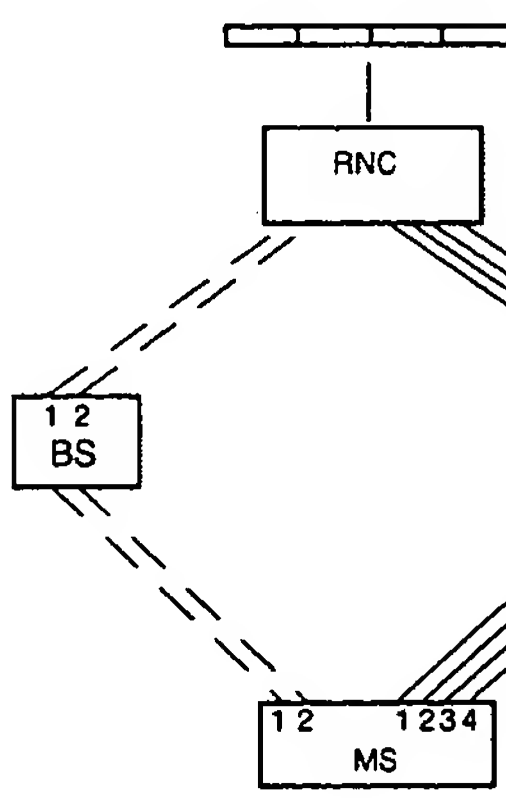

Figure 9. Macrodiversity in case of multibearer control at layer 2.

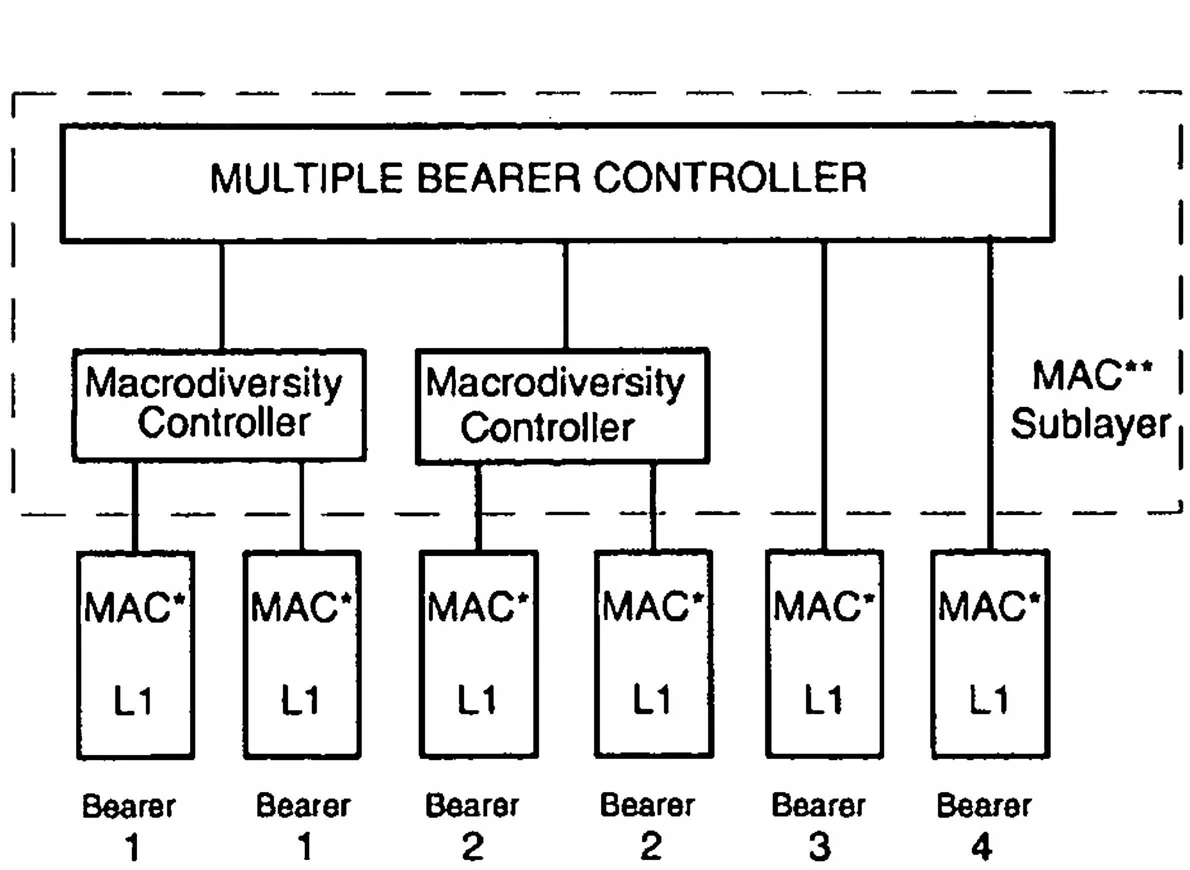

Figure 10. Allocation of multibearer control in the CODIT radio protocol stack.

Having independent bearers with independent channel coding simplifies very much the implementation of variable data rate services with one-bearer step size: they could simply be handled by the multibearer controller entity upon the requests for changing the data rate coming from the upper layers. In a similar way also the temporary loss of some bearers can be easily handled by the multibearer controller, informing the upper layers that only a reduced service can be provided, but mantaining the connection (this case should arise during handover phases where some bearers could be lost in the transition process from one BS to another one). Finally, also the provision of "asymmetrical services" with very huge differences between the data rate used on the uplink, with respect to the data rate used on the downlink, could be easily handled by a multibearer controller located in the MAC** sublayer managing independent bearers.

It should be noted that this solution does not allow the improving of the channel coding functionality by spreading the transmission errors, occurred on a single bearer, over the whole logical channel. A partial solution to this disadvantage could be provided by a similar spreading of adjacent bits, at the DLC frame level, over the different MAC connections, each of them having an independent channel coding. In such a way, the transmission errors, coming from one single bearer, could be spread over the DLC frame, possibly improving the DLC error protection mechanisms instead of the channel coding.

As a final remark, it should be noted that, if BSs belonging to different RNC groups are involved in the communication, some problems can arise. As a matter of fact, the proposed protocol architecture does not allow the use of BSs connected to different RNCs for a multibearer connection, this being possible only moving the multibearer control functionality to the DLC layer and then to the MCN. In such a way the multibearer connections should be called "multi DLC-connection links", allowing for several parallel DLC connections, one for each bearer, on the RNC-MCN interface. In this case all the multibearer connections would always be managed at the MCN level, also when the involved BSs are connected to the same RNC.

From the macrodiversity point of view, the current presence of the multibearer controller in the RNC allows the use of the DLC macrodiversity functionality, in the MCN, only if a complete multibearer connection (i.e., including all the involved bearers) is established within both the RNC groups (a multibearer connection is established within an RNC group if it is using any combination of bearers set up with BSs connected to the same RNC). As a matter of fact, the MCN must receive two different DLC connections from the two RNCs and one DLC connection can be established only if all the involved bearers can be received by the RNC. The choice of having a multi DLC-connection controller in the MCN, introducing a correspondence between each bearer and one single independent DLC connection,

could allow more flexibility, using the two level macrodiversity functionalities described above on each single bearer or DLC connection respectively, depending on the BS-RNC relative position (i.e., information coming from BSs connected to the same RNC are combined at the MAC layer in the RNC, while information coming from BSs connected to different RNCs are combined at the DLC layer in the MCN). Only after all these combinations the multiple DLC-connections are joined together within the MCN providing the layer 3 with only one logical channel.

4.2.3. Variable rate control

The most challenging approach, in defining the UMTS system concept, is to leave high flexibility, so that to allow the introduction of even today completely unknown services at every stage of the system life time.

DS-CDMA is felt to be well suited for the UMTS for a number of reasons. One reason is that it is intrinsically tailored to variable-rate information streams. The rationale in considering the variable-rate facility for such a system is that the information associated with the basic speech service (speech and signalling) is time-varying. In particular we are here referring to a Fast Variable-Rate (FVR) concept, that is the actual bit-rate of each source can be controlled on a frame by frame basis, being the frame length in the order of 10 ms. Moreover, the channel conditions in a mobile connections are also time-varying, suggesting an optimum allocation of channel coding driven by channel measurements, both for speech and signalling.

It is reasonable to combine all these variable-rate input sources in the same architecture, considering a unique variable-rate control unit [8] concentrating all the intelligence and enabling global optimisation of the physical channel resources. One further facility permitted by this approach is to allow the needs of the system capacity to be taken into account. Whenever the traffic from other users is increasing, the maximum allowed bit-rate can be reduced or constraints on the average bit-rate can be set by the control unit.

In a mobile telecommunications system based on a DS-CDMA radio interface, since the system capacity is tied to the average interference generated by active users, the use of a variable rate transmission scheme is fundamental in order to completely exploit the available radio spectrum (see section 3).

As mentioned in section 3, the transmission on CODIT radio interface is organised into fixed periods of time called "frames": the information mapped onto each frame is instead called "physical packet". Therefore the easiest way to implement a fast variable rate scheme is to vary, on a frame by frame basis, the length of the physical packet to be transmitted.

The information that is communicated through speech is a highly time-varying variable. Several aspects contribute to this: the rate needed to transmit the information itself, the identity and emotional state of the speaker, as well as the rate of speaking. The irregular distribution of this information with time makes the variable bit-rate approach particularly appealing. At the same time, the signalling control channel associated to the traffic channel has to convey information with various nature, ranging from "slow rate" and predictable, i.e., sent on a regular basis, measurement results, to "fast rate" and unpredictable handover signalling. Therefore the most natural approach seems to be a variable bit-rate associated control channel having a high grade of variability, i.e., with the rate controlled on a frame by frame basis.

In CODIT system, the information streams generated by the two different sources (i.e., speech and signalling) and considered after their own specific channel coding, are multiplexed on the same physical channel. The global information stream, resulting after the multiplexing of speech and control data, can exceed or not the maximum bit rate allowed on a single physical channel, where this limit is set taking into account the minimum spreading gain required by the overall system performance (see section 3). The maximum allowed bit rate can be seen, on a frame basis, as the maximum length of the related physical packet. If this length is exceeded, different actions could be performed, in order to counteract the resulting reduction in the quality of service offered to the user. On one hand, the transmitted power could be increased, thus affecting the overall system capacity. On the other hand, a feedback action could be performed onto either the information sources or the related channel coders in order to decrease, in a controlled way, the requested bit rates, thus affecting the complexity of the transmission scheme.

The management of the variable rate is performed in the Variable Rate Control Unit (VRCU). This unit contains the intelligence of the subsystem and, based on the different information, allocates the actual rate to each block. A breakdown of the blocks involved and of the signals exchanged is shown in figure 11. Each source (speech and control signalling) delivers an information flow $r(n)$, which is a function of the frame n, to its dedicated channel coder module. After having introduced the proper redundancy, the bit-streams with gross rates $R_s(n)$ and $R_c(n)$ are multiplexed in the output bit-stream with rate $R_{tot}(n)$.

The VRCU has complete access to the information relevant to the best allocation of resources. The input data to the VRCU can be identified as follows:

- *Speech needs.* The speech coder analyses the input speech on a frame by frame basis and decides which is the rate most suited for that particular segment of speech.
- *Control needs.* The information on the amount of the control needed in the frame is passed to the VRCU.
- *Channel measurements.* The channel characteristics, which can be represented in terms of the expected bit error rate, determine the requirements for protection of the information. This information is also passed to the VRCU.

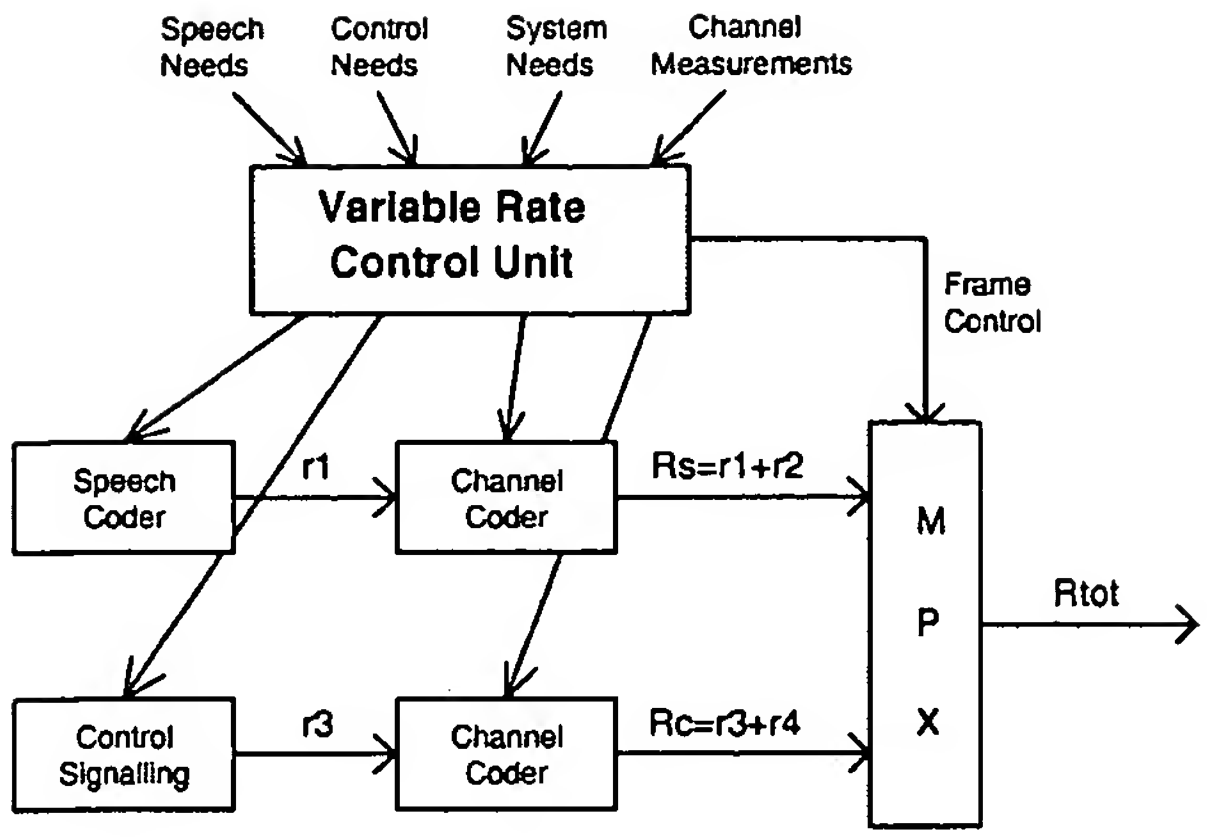

Figure 11. Block diagram of the units involved in the variable rate management.

- *System needs.* The traffic over the channel can change as a function of time as well as of the location of the mobile in the cellular environment. Different conditions can allow to use a higher rate for speech, with an associated better quality of conversation (e.g., by using wide-band speech coding), or can put constraints on the maximum possible data rate. This information relevant to the traffic from other users, shall be also made available to the VRCU.

All these input signals are considered in a proper algorithm in the VRCU that, as a result of an optimisation process, will deliver commands to each block. These commands are essentially the allocated rates and the allocated degree of protection. The algorithm that has to be implemented in the VRCU must weight all the different contributions to the total rate and cope with the constraint of maintaining the rate $R_{tot}(n)$ below the maximum capacity of the channel.

5. Conclusions

This paper shows how the main requirements of an advanced mobile communication system (e.g., high flexibility, support of variable bit rate services, high spectral efficiency), can be satisfied by a CDMA based radio interface. In particular, the paper focuses on the DS-CDMA radio interface designed within the RACE-II CODIT project, describing some of the technical solutions proposed for the provision of advanced features such as macrodiversity, multibearer transmission and variable bit rate services.

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